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(54) Method to perform link adaptation in enhanced cellular communication systems with several modulation and coding schemes

(57) Method to perform link adaptation at the radio interfaces of an enhanced packet data cellular network handling several Modulation and Coding Schemes (MCS) for maximizing data throughput. In a preliminary off-line step behavior in terms of net throughput of the various available MCSs is simulated for different C/I conditions. From the simulation two sets of tables are obtained, each table including upgrade and downgrade thresholds expressed in terms of Block Error Rate (BLER). Thresholds correspond to switching points from an MCS to the two available MCSs having the immediate less or more protection. The two sets of tables are referred to higher or lower diversity RF environments and are further specialized for taking into account EGPRS type II hybrid ARQ, namely Incremental Redundancy (IR). During transmission the transmitted blocks are checked for FEC and the results are sent to the network. The network continuously updates BLER using exponential smoothing. In order to achieve the correct time response, in spite of that RLC blocks can be received or not, a reliability filter is provided whose output is used to decide the weight between the new and old measurements to make the BLER filter impulse response exponentially decreasing with time. The IR efficiency is tested for each incoming block and an indicative variable IR_status is filtered using the same approach used for BLER. Each actual threshold of BLER to be used in link adaptation is obtained by a linear interpolation between the tabulated threshold without IR and with perfect IR, both weighed with filtered IR_status. Filtered BLER is then compared with said interpolated thresholds for testing the incoming of a MCS switching condition. Power control pursues the goal of maintaining constant QoS peak throughput per time slot.

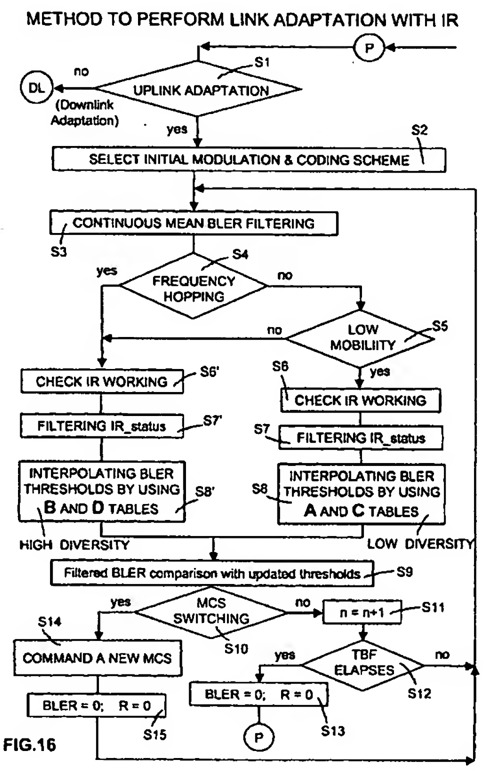

FIG.16

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EP 1 255 368 A1

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Description

FIELD OF THE INVENTION

[0001] The present invention relates to the field of radiomobile communication systems and more precisely to a method to perform link adaptation in enhanced cellular communication systems with several modulation and coding schemes.

BACKGROUND ART

[0002] A significant example of known art in the above technical field is disclosed in patent application WO 99/12304 filed by ERICSSON, titled: A METHOD FOR SELECTING A COMBINATION OF MODULATION AND CHANNEL CODING SCHEMES IN A DIGITAL COMMUNICATION SYSTEM". Both the invention disclosed in the cited document and the invention in subject, that will be disclosed later on, are suitable to be employed in the so-called General Packet Radio Service (GPRS) recently added to the Global System for Mobile communications (GSM) for enabling it to manage packet data. So an introduction of the GSM-GPRS system is needed before discussing the apparently nearest prior art. The introduction takes advantage from the large **GSM** standardization coming from ETSI (European Telecommunications Standards Institute) and also from the volume titled: "The GSM System for Mobile Communication", edited by the authors: Michel MOULY, Marie-Bernadette PAUTET, Copyright 1992).

[0003] **Fig.1** of the description is similar to the fig. 2 of standard ETSI **GSM 03.60 -** Service description. The system of fig.1 represents a cellular GSM(DCS)-GPRS(Enhanced) network including mobile stations communicating via radio with a fixed remaining part. In fig.1 are visible first type of Mobile Stations MS suitable for voice communication (and short messages) and second type of mobile stations named User Equipment UE each comprised of a Terminal Equipment TE for handling data (as a PC) connected to a Mobile Terminating equipment MT suitable to data packet transmission. Mobile stations MS and UE camped on a cell are connected via standard on air interface Um to a fixed Base Transceiver Station BTS which serves either a central or trisectorial cell belongs to a clustered geographical area covered by GSM-GPRS Public Land Mobile Network PLMN. In fig.1 more base stations BTS are connected to a Base Station Controller BSC through a not fully standardized Abis interface. The BSC controller includes a block PCU (Packet Control Unit) relevant for the present invention. The BSC controller and the interconnected base station BTS constitute a Base Station Subsystem BSS serving a cluster of cells. An BSC controller in its turn is connected to a Message Switching Centre MSC and to a Service GPRS Support Node SGSN via standard interfaces A and Gb respectively, both supporting SS7 signalling. The MSC centre and SGSN node are connected to a Home Location Register HLR and a Visitor Location Register VLR which add intelligence to the network by allowing mobility of communications. The MSC centre and SGSN node support Short Message Service SMS, being for this purpose connected to a Short Message Service Centre SM-SC via the functions SMS-GMSC (Short Message Service - Gateway MSC) and SMS-IWMSC (SMS - InterWorking MSC). The SGSN node is further connected to: 1) another SGSN node of the same PLMN network through a standard Gn interface; 2) a Gateway GSN node GGSN belonging to another PLMN network through a standard Gp interface; 3) a Gateway GSN node GGSN belonging to the same PLMN network, through the Gn interface, and the GGSN node is connected to either an IP (Internet Protocol) or X.25 Public Data Network PDN specialized in packet data routing; 4) finally to an Equipment Identity Register EIR. The MSC centre is connected to the Public Switching. Telephone Network PSTN also comprised of an Integrated Services Digital Network ISDN. Besides the mentioned interfaces also the following standard ones are provided: Gf, Gs, Gr, Gd, D, E, C whose connections are visible in fig.1.

[0004] The schematized GSM-GPRS system is capable to switch both the traditional voice and data circuits and the new packet data which don't request fixed connections for all the duration of an active session. The SGSN node has the same role for packet data as the MSC centre has for voice circuits, it traces individual locations of the mobile stations enabled for data packet communication and performs security and access control functions. For this purpose the HLR register includes information concerning GPRS users. The GGSN node provides interworking with external data packet switching networks, in particular with a backbone network based on IP protocol.

[0005] Both GSM and GPRS use standard procedures at the relevant interfaces, namely for: synchronization, cell selection and reselection, paging, access control, request a dedicated channel, security, error detection and correction, retransmission, power control, voice and data flux control, routing, handover, billing, etc. Such procedures belong to a most general protocol having a layered structure named "Transmission Plane" proposed by the International Organization for Standardization (ISO) for Open System Interconnection (OSI). Based on ISO model an OSI system can be described by means of a set of subsystems fit in a protocol stack. A subsystem N which consists of one or more entities of level N interacts only with subsystems immediately upon and below it and a level N entity operates into its own level N. Peer level N entities communicate each other by using services from the underlying layer N. Similarly, layer N services are provided to the layer N+1 at an N-Service Access Point named N-SAP. Information transferred from a starting to an arrival point is always conveyed by physical channels provided at the crossed interfaces. Relevant layers

for the arguments developed in this disclosure are the following:

- Radio Link Control / Medium Access Control (RLC/MAC). The RLC layer-2 function provides a radio link with reliability and maps into GSM physical channels the Link Layer Control (LLC) layer-3 frames. The MAC function is provided to control and signalling procedures for accessing radio channel, i.e. request and grant. RLC/MAC protocol is standardized in **GSM 04.60.**
- GSM RF is pertaining to the physical radio channel at the Um interface as standardized in the series of specifications **GSM 05.xx** The physical channel relevant for GPRS service is named PDCH (Packet Data Channel).

[0006] At GPRS planning stage the compatibility with pre-existent GSM has been deliberately maintained to enable GPRS of exploiting the same physical channels as GSM at the Um interface and consequently promoting an easy integration. Both for GSM and GPRS there are signalling channels and traffic channels, the first ones are either for broadcast common control or for dedicated control, the second ones are either for voice or packet data. The additional logical GPRS channels, although referred to packet data have names and functional characteristics which follow from the conventional GSM channels; examples of relevant GPRS channels are the following: PBCCH (Packet Broadcast Control Channel), PCCCH (Packet Common Control Channel), PACCH (Packet Associated Control Channel), e PDTCH (Packet Data Traffic Channel). A list of relevant channels is reported in the specification **GSM 05.01** titled "Physical layer on the radio path".

[0007] The Extended GSM 900 system is required to operate in the following frequency bands:

- 880 - 915 MHz: mobile stations transmit uplink, base station receives;
- 925 - 960 MHz: base station transmits downlink, mobile stations receive; while for Digital Cellular System DCS 1 800 the system is required to operate in the following frequency bands:
- 1.710 - 1.785 MHz: mobile stations transmit uplink, base station receives;
- 1.805 - 1.880 MHz: base station transmits downlink, mobile stations receive.

Each of the above frequency band is also used in GPRS service and includes a plurality of modulated carriers spaced 200 kHz apart. Full-duplex communications take place by Frequency Division Duplexing (FDD) technique. A carrier among those in use in a cell is assigned for all the duration of a timeslot TS out of eight cyclically repeated to allow time division among the users. During the assigned timeslot Modulation (detailed in **GSM 05.04)** impresses the characteristics of the modulating burst onto one or more physical parameters of a digital carrier to be transmitted at radiofrequency. The GSM-GPRS system exploits a GMSK (Gaussian Minimum Shift Keying) modulation that is a non-linear Continuous Phase Modulation (CPM) characterized by compact spectrum and constant modulation envelope. Compact spectrum generates poor interferences into adjacent frequency channels by introducing a slight worsening of the intersymbolic interference. Constant modulation envelope allows the gain saturation of the power amplifier (class C amplifying) and consequent energy saving from the power supply. Besides power control becomes simpler.

[0008] With reference to **fig.2** it can be appreciate the sequential organization of 8 timeslots TS0, ..., TS7 constituting a 4,615 ms basic frame used in Time Division Multiple Access (TDMA) GSM-GPRS system. Four different typologies of burst are provided corresponding to the possible contents of any timeslot. The sequential frames are organized within more hierarchical levels observed by all the carriers used in the system. All the carriers transmitted by a BTS have reciprocally synchronized frames. Starting in the figure from bottom to top each timeslot has 0,577 ms duration, corresponding to 156,25 $\times$ 3.69 µs bit duration, and carries an information burst containing 142 useful bits, 3+3 tail bits TB, and a guard time GP without information 8,25 bits long. The 3.69 µs bit duration corresponds to 270,83 kbit/s which is the system cipher rate. The burst can be of four different types, namely: Normal burst, Frequency Correction burst, Synchronization burst, and Access burst. For the purposes of disclosure the only Normal burst is depicted in fig. 2 where it includes 2$\times$58 useful bits, redundancy included, and 26 bits of a training sequence in midamble position. Training sequence is a known pattern used to dynamically synchronize the received burst and to estimate the impulse response of the radio channel for correctly demodulating the incoming signal. The nature of the 116 bits payload will be detailed later on, distinguishing between GSM and GPRS. Continuing towards the upper part of **fig.2** it can be noticed that two different typologies of multiframes are foreseen, namely a signalling multiframe for carrying control channels and a traffic multiframe for carrying payloads and associated signalling. The signalling multiframe is 253,38 ms long and includes 51 basic TDMA frames. A GSM traffic multiframe is 120 ms long and includes 26 basic TDMA frames. A GPRS traffic multiframe is 240 ms long and includes 52 basic TDMA frames. The two type of multiframes concur to form a unique superframe 6,12 seconds long, consisting of 1326 basic TDMA frames, finally 2048 sequential superframes form one iperframe of 2.715.648 basic frames TDMA of 3h 28m 63s 760ms duration. A frame Number FN referred to the frame position in the iperframe is broadcasted within the cell.

[0009] **Figures 3a and 3b** show traffic channel organization in the TDMA multiframes for voice/data and packet data respectively. **Fig.3a** concerns GSM payload where a multiframe of 26 basic frame includes: 24 traffic frames (T), 1

associated control frame (A), and 1 idle frame (-). A physical channel inside a multiframe is constituted by the combination of one frequency and one repetitive time slot. A burst of fig.

[0010] 2 generates a period of RF carrier which is modulated by the relevant data stream. A burst therefore represents the physical content of a timeslot.

[0011] **Fig.3b** concerns GPRS payload where a multiframe of 52 basic frame includes 12 radio blocks B0, ... B11 of 4 basic frames each, intercalated with an idle frame (X) every three radio blocks. A radio block is carried on a channel defined as above spanning over 4 TDMA frames, so as the mean transmission time of a RLC block is near 20 ms.

[0012] **Fig.4** is referred to the GPRS service and shows a mapping of sequential RLC layer blocks into physical layer. Each RLC block includes a block header BH of variable length, an information field comprising data coming from the upper layer LLC, and a field Block Check Sequence BCS used for error detection. A single RLC block is mapped into 4 sequential frames of the TDMA multiframe. So until 8 users can be interleaved in the period of a radio block.

[0013] GSM's payload timeslots are allocated one to one to the different users, both in uplink and downlink, while as far as concerns GPRS service a flexible allocation is available. More precisely: 1) GPRS's payload timeslots are independently allocated in uplink and/or downlink; 2) singular users can take advantage of multislot allocation; 3) each configured data packet physical channel PDCH (timeslot) can be advantageously shared among different users which access it on the basis of appropriate priority rules. The MAC layer of GPRS protocol has appropriate procedures for governing dynamic allocation of the resources. Control messages to set up or set down a connection activate said procedures for packet data transfer. Temporary Block Flows (TBF) are connections set up on physical layer by the MAC procedures, they include memory buffers to accommodate the queues of RLC/MAC radio blocks. Each TBF connection allows unidirectional point-to-point transfer of user data and signalling between a mobile station and base station, or vice versa. A TBF connection is held for the only transfer of all the RLC/MAC blocks of a LLC protocol session. The network assigns to each TBF connection a respective Temporary Flow Identity named TFI identifier by associating a field in the header of RLC/MAC blocks. The mobile stations shall assume that TFI identifier is unique for uplink or downlink concurrent TBFs (i.e. assigned to the same MS/UE). The header of RLC/MAC blocks further includes fields to specify direction and type of a control message.

[0014] In case of dynamic allocation of the resources and in presence of at least one uplink TBF connection, the header of each RLC/MAC block transmitted downlink includes an Uplink State Flag field (3 bits) named USF written from the network to enable the uplink transmission of a successive radio block from one out M mobile stations which share the same uplink PDCH channel.

[0015] GSM-GPRS system bears three classes of operation for mobile stations: a class A mobile operates with GSM and GPRS simultaneously; a class B mobile watches GSM and GPRS control channels but can operate only a set of service at a time; finally a class C mobile only uses GPRS services. Furthermore physical resources at the Um interface can be shared between speech and packet data services on the basis of traffic charge at the initial cell planning.

[0016] GPRS service bears Quality of Service (QoS) to assure among other things the following requirements: respect of a negotiated priority of service, service reliability, guarantee of a fixed end-to-end packet transfer delay, guarantee of medium and peak throughput in conformity with a certain multi-slot class. QoS parameters together with A, B, and C classes of operation and class of multislot capability take part in a User Profile made known to the network during GPRS attach.

[0017] A generic cellular telephony system suffers a lot of impairments mainly due to the following causes:

1. The peculiarity of radio propagation in conformity with the typology of the cells.
2. The mobility of the users
3. The intrinsic frequency reuse.

[0018] An impairment due to the first cause of above is the time dispersive behavior of the propagation medium because of non-linearities which distort the original shape of transmitted pulses, causing intersymbol interference due to the pulse spreading over adjacent symbol intervals. Another impairment descending from the same cause is undoubtedly multipath fading due to the random presence of spotted atmospheric diffusers on the radio path introducing statistical behavior on the radio propagation. Both at MS or BTS receiving antennas, various phase-shifted echoes of a transmitted signal coming from multipath are summed up with random distributed phases. The result is an amplitude envelope attenuated below certain levels during corresponding fade durations taken as observation times. The time-varying fading behavior is a statistical process whose probability density follows the Rayleigh distribution. Multipath fading is spectrally characterized to be either flat or frequency selective (notch), this happen respectively for correlated or uncorrelated scattering and in both the cases it generates burst errors. Last shortcoming of an on air interface is its vulnerability due to the easiness of malicious interception of data and conversations, if not otherwise provided.

[0019] Impairments due to the user mobility mainly are: shadow fading (i.e. corner), Doppler effect, and a certain spreading of Time Of Arrivals (TOA) of RF signals at the BTS antenna because of the various distances of the mobile

stations. Shadow fading is caused by the incoming of an obstacle along the line of sight propagation. By shadow fading the transmitted RF signal undergoes an additional steep attenuation to the usual path attenuation. Doppler effect is a slight frequency shift proportional to the speed the mobile station; the shift introduces noise phase and makes time-variant the channel response. Doppler effect for the highest speeds disturbs the synchronization process and the estimate of the channel pulse response consequently. The spreading of the TOAs force the realignment of the RF signals at the BTS side.

[0020] An impairment due to the frequency reuse is the presence of isofrequential interferent signals coming from the neighbor cells. C/I ratio increases and the quality of the reconstructed signal gets worsen consequently. The smaller are the cells the greater is the allowed traffic throughput but parallely cochannel interference increases due to the heavy frequency reuse.

[0021] The finding of strategies effective to neutralize the above causes of misoperation in cellular systems, needs a good knowledge of the various electromagnetic environments. Typical Radiomobile channels have been extensively studied and experimented with large, medium, and small cells. Small urban cells are further subdivided into micro and pico cells (i.e. canyons). Large and medium cells are subjected to a variety of environments, such as: hilly terrain, mountainous, woody, motorway, urban, etc. Starting from the above considerations ETSI standard committee has specified in **GSM 05.05** some practical pulse responses for typical radiomobile channels, such as: Hata-Okumura, COST231, TU3, TH, etc. Most recently models have been proposed in which the RF channel is also spatially characterized.

[0022] A lot of countermeasures have been introduced into the cellular systems to combat the above drawbacks; the most popular are the following: channel coding - interleaving - ciphering - channel equalization - frame alignment - frame (block) retransmission - slow frequency hopping - power control - intra-cell handover - and lastly link adaptation. They are valid in general so that speech, traffic data, and signalling can take advantage from them. Obviously link adaptation is the countermeasure that mostly impacts the present invention: it can be specialized into speech or data adaptation. Recently link adaptation has been improved in concomitance with the GSM enhancement, but before introducing link adaptation the older countermeasures will be considered.

- Channel coding introduces redundancy into the data flow increasing its rate by adding information calculated from the source data in order to allow the detection or even the correction of signal errors introduced during transmission. The result of the channel coding is a flow of code words (i.e. blocks as far as concerns block coding). In the case of speech, for example, blocks of 260 bits each are generated every 20 ms at the output of the 13 kbit/s voice encoder. Block coding with parity and convolutional codes, well detailed in **GSM 05.03** introduce redundancy increasing the bits from 260 to 456. Coding schemes make generally use of Puncturing Schemes (PS) acting on block convolutional codes for keeping only q bits out of *pn* through a pre-determined rule. Puncturing permits to reach an efficiency ratio (ratio between the number of useful bits in the source sequence and the number of bits actually transmitted) which is limited to fractions of the form *p/q*, otherwise impossible without puncturing. Parity code adds parity bits to the bits to be convolutionally coded for checking the failure of block convolutional code in error correction. For the sake of completeness a so-called Fire code is prevalently used in fast signalling channel bursts (in-band FACCH) and BCS Header field of GPRS. Fire code is a gender of cyclic code which adds redundancy dedicated to the detection and correction of "bursty" errors. Since a block convolutional code is mainly used for error correction and error often come out in group, the fire code is used in concatenation and noticeably improves the decoded information. Each type of channel has its own characteristic coding scheme. Channel decoding is performed through a de-convolution process which takes advantage from "soft decisions" delivered from the demodulator. Soft decision is an estimated probability of correctness of each detected bit. A convolutional decoder based on the Viterbi algorithm simply exploits Euclidean metrics to implement soft decisions.
- Interleaving consists in mixing up the bits of several code words (code blocks), so that bits which are close to one another in the modulated signal are spread over several code words. Since the error probability of successive bits in the modulated stream is very much correlated, and since channel coding performance is better when errors are de-correlated, interleaving aims at de-correlating errors and their position in code words. In the case of speech, the preceding 456 code bits are reordered and partitioned and diagonal interleaved with 8 timeslot depth to spread the burst errors over more bursts maintaining a reasonable delay of about 37,5 ms (65 burst .periods). De-interleaving is the opposite operation.
- Ciphering modifies the content of a code block through a secret recipe known only by the mobile station and BTS station. The original content (2×57 bit semi-bursts) is encrypted by summing bit by bit to a ciphering flow. Deciphering is the opposite operation. Ciphered coded blocks are differentially encoded before modulation to prevent error propagation.
- Frame alignment takes advantage from Burst formatting which adds some binary information to the ciphered code blocks of 2×57 bit semi-bursts in order to help synchronization and equalization of the received signal and fast signalling. Fig.2 shows that the added information include: 26 bit training sequence, 3+3 TB tail bits, and 1 stealing

flag bit for each 57 bit semi-burst (total 8 bits for the 20 ms speech block) indicating either the semi-burst contains user data or is used in fast associated signalling mode (FACCH). The transmitted training sequence (known to the receiver) has a central peak in its autocorrelation function whose detection from the receiver allows the burst synchronization. Frame alignment is governed by the BTS which measures the TOAs of all the received RF bursts and sends to each mobile station a rêspective command forcing a delay in the start of transmission in order to maintain constant three frame offset between uplink and downlink bursts.

- Channel equalization usually tempts to reshape the received pulses in order to reduce the intersymbol interference before the demodulation. Contrarily to this definition, an equalizer based on Maximum Likelihood Sequence Estimation (MLSE) criteria, as that based on the Viterbi algorithm, doesn't attempt to equalize the channel in strict sense, but rather uses the knowledge of the channel pulse response (get from the training sequence estimation) to find the data sequence transmitted with the maximum probability. In this area the most recent techniques use beamforming for estimating space and time channel responses. This allows to position the most incoming RF energy towards the directions of the useful signal and its echoes, to the detriment of cochannel interferents. The result is an optimized channel pulse response.
- Block retransmission under Automatic Repeat Request (ARQ) scheme when a code block (different from speech) undergoes one or more residual errors.
- Slow Frequency Hopping (SFH) is a gender of frequency diversity technique descending from the aptitude of Rayleigh fading to be uncorrelated with frequencies spaced sufficiently apart: i.e. 1 MHz. SFH is the interchangeability of the carriers assigned to the physical channels timeslot by timeslot. SFH is carried out inside an orthogonal set of frequencies in use into a cell; the hops are matched between MSs and BTS because of FDD duplexing. For this aim the system refers to a hopping sequence generation algorithm (detailed in **GSM 05.02)** which uses an index MAIO (Mobile Allocation Index Offset) linked to the Frame Number FN.
- Power control (detailed in **GSM 05.08)** is a BSS procedure which step by step modifies, within some range, the uplink/downlink RF transmission power. Power control is based on SACCH Measurement Result message and remedies for path loss and shadow attenuations, further improving spectral efficiency by reducing the overall interference of the system. Secondly it extends battery life of the mobile stations.
- Intra-cell handover (detailed in **GSM 05.08)** is a particular case of the handover procedure charged to switch the mobile station on a free channel of the same cell when transmission quality drops below a given threshold. If an intra-cell handover is successfully the radio link failure can be avoided.

[0023] As already outlined, GPRS service has been added to the GSM in order to achieve higher performance with data handling. The introduction of packet switching capability meets this objective. **TABLE 1 of APPENDIX 1** shows four standard GPRS coding schemes CS-1 to CS-4 relevant to a RLC block. One block of 456 coded bits carries one radio block. CS-1 consists of a half rate convolutional code for FEC and a 40 bit FIRE code for BCS (and optionally FEC). CS-2 and CS-3 are punctured versions of the same half rate convolutional code as CS-1 for FEC. CS-4 has no FEC. Traffic channels exploit CS-1 to CS-4 while signalling channels prefer CS-1. Practical data-rates (kbit/s) achievable on a single GPRS time-slot are shown in the last column of Table 1.

[0024] A subsequent goal of GPRS specifications has been that to increase the data-rate. This aim has been reached by an Enhanced GPRS (EGPRS) version characterized by a higher modulation level, namely 8-PSK (Phase Shift Keying) in combination with additional five coding schemes. In case of 8-PSK modulation a block of 1368 coded bits (456 coded symbols) carries one radio block. While the only GMSK modulation allows to the GPRS users a theoretical bit-rate spanning between 9 and 150 kbit/s (the higher bit-rate being obtained with the poor coding scheme CS-4 and all the eight available time-slots), the 8-PSK modulation allows to the EGPRS users a theoretical bit-rate until 450 kbit/s, triplicating the previous one. In the new EGPRS context, because of the choice between two type of modulations, namely GMSK and 8-PSK, an assignment message shall specify both Modulation and Code type assigned to the channel. Nine combinations of Modulation and Coding Schemes, MCS-1 to MCS-9, are foreseen and detailed in **GSM 05.03, GSM 05.04,** and **GSM 04.60.**

[0025] **TABLE 2 of APPENDIX 1** shows: code rate, data rate, number of coding bits, etc. concerning EGPRS MCS-1 to MCS-9 schemes. In TABLE 2 the column HCS means Header Check Sequence, while the column Family will be explained later. New EGPRS service thanks to the nine MCSi combinations offers several more opportunities for packet data link adaptation. From **TABLE 2** it can be observed that for each type of modulation the grater the code-rate, the greater is the data-rate, because code-rate represents the ratio between the number of useful bits in the source sequence and the number of coded bits. Considering burst having fixed length it results that the higher the code-rate, the poorest is the protection against errors. Higher level modulations (like 8-PSK) are more sensible than lower level modulations (like GMSK) to the causes of RF link degradation and similarly higher code-rates in comparison with lower code rates. Greater sensibility also means faster worsening of the signal delivered to the users as the quality of the RF link worsen. Nevertheless the enhanced opportunity to select one out several combinations of modulation and coding schemes (MCS), enables the system to switch among the various MCSs during run-time to combat the variability

of the RF channel. Link adaptation is just this behavior! TABLE 2 doesn't limit the present invention which is valid also in presence of different high level modulations variously combined with the same or different coding schemes.

[0026] Link adaptation oriented to voice services promotes speech quality compatibly with the variable conditions of the RF link; on the contrary link adaptation oriented to packet data services promotes higher throughputs. In both cases a compromise between data-rate and quality of transmission shall be inevitably pursued when selecting a new modulation and coding scheme. Quality of transmission, and more in general quality of service, plays a main part in a radiomobile system which normally attempts to optimize its own operation by constantly monitoring a lot of parameters. For this aim a variety of measures directed to uplink and downlink transmissions are usually performed, such as (with reference to the single MS): delay of synchronization, channel pulse response, power level of the modulated carrier, power level of the interferent signals, carrier to interferences power ratio (C/I), signal to noise power ratio (S/N), Bit Error Rate (BER), Bit Error Probability (BEP), etc. Incoming useful and interferent signals from the neighbor cells are even monitored to compile a candidate list for handover. The measures performed by the mobile stations are joined to those performed directly by the BTS and sent forward to the BSC to enable its control capability in the opposite direction. The performed measures give support to the most known procedures of the radiomobile system, such as: Cell selection and reselection, Timing advance, Power control, Handover, link adaptation, etc.

[0027] Henceforth packet data transmission will be only considered, because voice/circuit data link adaptation is not particularly relevant for the invention in subject. Consequently the remaining part of the disclosure will be preferably referred to the GPRS/EGPRS improvement of the GSM. Decisions concerning link adaptation for packet data shall be inevitably issued from a high level protocol agent having the supervision of the signalling conveyed through the uplink TBFs and the opportunity to send command through downlink TBFs. The PCU functional block of fig.1 represents a unit charged to manage RLC/MAC blocks and consequently take high level decision about link adaptation. Two different modes of operation are foreseen in the RLC/MAC protocol: acknowledged mode and non-acknowledged mode.

- Acknowledged mode (non-transparent service). Transfer of RLC Data Blocks in the GPRS acknowledged RLC/MAC mode is controlled by a selective ARQ mechanism coupled with the numbering of the RLC Data Blocks participating a Temporary Block Flow. The sending side (the MS or the network) transmits radio blocks within a window and the receiving side sends either Packet Uplink Ack/Nack or Packet Downlink Ack/Nack message when needed. Every such message acknowledges all correctly received RLC Data Blocks up to an indicated block sequence number (BSN), thus "moving" the beginning of the sending window on the sending side. Additionally, a bitmap that starts at the same RLC Data Block is used to selectively request erroneously received RLC Data Blocks for retransmission. The sending side then retransmits the erroneous RLC Data Blocks, eventually resulting in further sliding the sending window. The RLC acknowledged mode shall be used for data applications where the payload content needs to be preserved. It will be the typical mode for Background class (background delivery of e-mails, SMS, download of databases) and Interactive class applications (web browsing). In EGPRS TBF the transfer of RLC Data Blocks in the acknowledged RLC/MAC mode can be controlled by a selective type I ARQ mechanism, or by type II hybrid ARQ mechanism dealing with Incremental Redundancy (IR), both coupled with the numbering of the RLC Data Blocks within one Temporary Block Flow. In the type I ARQ mode, decoding of an RLC Data Block is solely based on the prevailing transmission (i.e. erroneous blocks are not stored). In the type II hybrid ARQ case, erroneous blocks are stored by the receiver and a joint decoding with new transmissions concerning original blocks is done. If the memory for IR operation run out in the MS, the MS shall indicate this by setting an LA/IR bit in the EGPRS PACKET DOWNLINK ACK/NACK message. Type II hybrid ARQ is mandatory in EGPRS MS receivers.
- Non-acknowledged mode (transparent service). The transfer of RLC Data Blocks in the unacknowledged RLC/MAC mode is controlled by the numbering of the RLC Data Blocks participating one Temporary Block Flow, but it does not include any retransmission. The receiving side extracts user data from the received RLC Data Blocks and attempts to preserve the user information length by replacing missing RLC Data Blocks by dummy information bits. Delay sensitive services, such as Conversational class (voice, video conference) and Streaming class applications (one-way real time audio and video) will make use of the RLC unacknowledged mode. The same mechanism and message format for sending temporary acknowledgement messages is used as for acknowledged mode in order to convey the necessary control signalling (e.g. monitoring of channel quality for downlink channel, or timing advance correction for uplink transfers). The sending side (the MS or the network) transmits a number of radio blocks and then polls the receiving side to send an acknowledgement message. A missing acknowledgement message is not critical and a new one can be obtained whenever.

[0028] Quality of Service (QoS), see **GSM 03.60,** takes advantage from both transparent or non-transparent transmissions, as indicated for the services listed above. The two transmission modes differently impact the two QoS classes concerning point-to-point delay and throughput. Unacknowledged packed data is characterized by a fixed point-to-point delay and a variable gross bit-rate, mainly due to the system attempts to maintain a target user bit-rate with the

required quality. On the contrary, due to retransmissions, acknowledged packed data is characterized by a variable point-to-point delay and a variable user bit-rate which can be calculated with the following known expression:

$$\text{Throughput}_{NET} = \text{Throughput}_{MAX}\,(1 - \text{BLER}) \qquad (1)$$

where: Throughput_{NET} is the net user bit-rate; Throughput_{MAX} is the peak user bit-rate; and BLER is the Block Error Rate on the current Modulation and Coding Scheme (MCS).

[0029] Link adaptation is applicable in packet data transmission for both the acknowledged and the unacknowledged transmission modes. Other questions about link adaptation are the following:

- compatibility of the link adaptation with power control;
- effect of frequency hopping on link adaptation;
- the effect of incremental redundancy.

These questions are briefly discussed in the following.

[0030] Both Link Adaptation and Power Control are features that aim at network optimization but, if run independently, may lead to a contrasting situation. Link Adaptation tries to optimize performance (i.e. maximize throughput) for a given radio link quality. This means that if, for instance, radio conditions are improved, the known methods of Link Adaptation try to benefit from this situation and increase the overall throughput by switching to a different (less protected) coding scheme. On the contrary Power Control tries to reduce interference and save power by using the least possible transmit power suitable to achieve a specified C/I ratio (which is consistent with a required minimum performance). In other words PC tends to keep constant the radio link quality thus inhibiting further improvements due to the LA algorithm. Therefore a common strategy has to be decided to make LA and PC work together.

[0031] Frequency hopping increases the variability of the channel so that the choice of an idoneous MCSi shall be conditioned consequently, for example channels having higher variability should require more robust MCSs and consequently lower throughputs.

[0032] Incremental redundancy pertaining to type II hybrid ARQ, differently from type I ARQ, needs a lot of memory to store erroneous block together with multi-bits soft decisions usable in joint decoding the successive retransmitted bits. The overflow probability of an IR buffer de facto increases with the less robust MCSs at the lowest C/I; when this happens lastly stored blocks are discarded and BLER starts to increase. The capacity to contrast worsening of the service clearly depends from the skill of link adaptation to manage this circumstance.

[0033] Now the attention is turned back to the patent application WO 99/12304 filed by ERICSSON whose claim 1 sounds like that: In a communication system, a method for selecting a combination of modulation and channel coding scheme from a plurality of combinations of modulation and channel coding schemes comprising the steps of:

- measuring at least one link quality parameter of an RF link (see claim 2: C/I, BER, received signal level, time dispersion; see claim 8: user data throughput; see claim 9: BLER);
- calculating at least one channel characteristic measure based on the measured at least one link quality parameter (see claim 3: variance; claim 4: mean value);
- estimating user quality values (see claims from 5 to 12) for each one of the combinations of modulation and channel coding schemes based on the calculated channel characteristic measure (variance, mean); and
- selecting a combination of modulation and channel coding schemes (MCSi) on an RF link that provides the best user quality value (see also claim 15: performed during idle states or wait states).

[0034] The step of estimating user quality values is well detailed in the dependent claims as far as it concerns: using of simulation results - using of laboratory results - run time estimating - estimating user data throughput - estimating BLock Error Rate (BLER) - estimating BLER and nominal bit-rate - mapping the calculated media or variance into BLER for each MCSi - estimating speech quality. The above claim doesn't explicitly mention the use of ARQ retransmission. The selected MCSi combination provides for the best user quality value. This claim appears mostly oriented to perform link adaptation for speech service or transparent data service.

[0035] Another independent method claim (16) adds to the preceding claim 1 the feature of: "communicating data using a non-transparent service over an RF link". Contextually user data throughput is estimated instead of quality and the selected MCSi combination provides for the best user data throughput. Clearly this claim appears mostly oriented to perform link adaptation for not-transparent packet data service.

[0036] Another feature added to the independent claims concerns the determination of an optimal transmit power level for each MCSi scheme previously selected by link adaptation. The optimal power is determined based on the

measured C/I and its level is limited by a dynamic range of the power transmitter.

[0037] Main purpose of this prior application is that of calculating the variance of the measured quality parameters, other than the usual averages considered in the oldest methods, for the precise aim to consider the variability of the RF channel when performing a dynamic link adaptation.

COMMENTS ON THIS PRIOR APPLICATION

[0038] Page 25 and Figure 9 of the cited document clarify the wording of the claims.
The clarified method referred to the packet data sounds like that:

1. (see block 112 of fig.9) Measure of some quality parameters (C/I, BER, etc.) and calculation of the relative mean value and variance.
2. (see block 114 of fig.9) Mean value and variance are sent to the inputs of some tables, or mapping functions, obtained from simulations, or through laboratory tests, etc. whose outputs (or mapped values) are the expected BLER(i) for all the MCS(i).
3. (see block 116 of fig.9) By using the know relation: T(i) = T_max(i)*(1-BLER(i)) the throughput T(i) obtainable for that specific condition of mean value and variance is calculated for each MCS(i).
4. (see block 118 of fig.9) In correspondence of the maximum throughput T(i) the respective MCS(i) is selected for coding and modulating the user signal.

[0039] An evident drawback is the intrinsic difficulty of the proposed method, mainly due to the following causes:

1. Time and efforts spent for collecting sufficient measures, further increased to the calculation of averages and variance of the measures (variance needs the knowledge of the average). Measure of C/I is not easy to do.
2. Additional signalling (in the only case of downlink adaptation) to transfer the calculated mean value and variance from the mobile station to the network. Nevertheless this is true only if the measures normally executed for power control and handover were not considered.
3. Off-line calculation of cumbersome tables (or mapping functions) for mapping mean values and variances of a measured parameter into corresponding BLERi for all the MCSi. That because each table of the type BLERi(C/I), or BLERi(BER), etc. shall foreseen entries for two variables and one output for providing BLERi, for example: BLERi(average C/I, variance). So the complete input has reasonably to keep into account several possible variances for each mean value considered in a significant grid.
4. The more complicated the mapping tables are, the more they suffer from sensitivity of the parameters, consequently the empirical representation of a certain BLERi requires to guess exact combinations of mean and variance.
5. Nothing is said on the link adaptation impact in case of non-transparent ARQ with incremental redundancy (type II Hybrid ARQ) implementation. The only use of non-transparent ARQ retransmission without incremental redundancy (type I ARQ) is mentioned in the text. Incremental redundancy impacts the off-line simulations and needs some expedients to be correctly implemented together link adaptation.
6. Optimal transmission power is dependent from the selected MCSi and requires the definition of as many C/I target as the MCSi.

[0040] Being the prior art document silent about point 5, it is not completely proved that the known method works satisfactory with Incremental Redundancy.

OBJECTS OF THE INVENTION

[0041] The main object of the present invention is that to remedy to the defects of the prior art and indicate a method for dynamically optimize data throughput at the radio interfaces of a packet data cellular network.

[0042] Other objects of the invention is that to optimize data throughput at the radio interfaces in presence of slow frequency hopping and/or high user mobility.

[0043] Other objects of the invention is that to optimize data throughput at the radio interfaces in presence of retransmission with incremental redundancy of bad received radio blocks.

[0044] Further objects of the invention is that to harmonize power control and link adaptation mechanisms jointly active at the radio interfaces.

SUMMARY AND ADVANTAGES OF THE INVENTION

[0045] To achieve said objects the subject of the present invention is a method for dynamically optimize data throughput at the radio interfaces of a packet data cellular network, as disclosed in claim 1.

[0046] The link adaptation method of the present invention directly calculates and continuously updates for the active uplink and/or downlink connections, the ratio between the not correctly received blocks and the transmitted blocks, namely the BLER, contrarily to the prior art method in which a BLER is acquired through cumbersome measuring and mapping step

[0047] s. The method of the present invention need not additional signalling for adaptation downlink, that because the number of not correctly decoded blocks constitutes the minimum information that a mobile station have to transmit to the network for requesting retransmission or for quality estimation.

[0048] Once the BLER has been calculated, the problem to be solved is that of how the BLERi for other MCSi would have been for the same channel conditions, in order to establish in base to the known relation (1): $Ti = Ti_{MAX}(1 - BLERi)$ if a change of MCS is needed towards one MCSi having higher throughput Ti. The novelty of the present invention is that relation (1) is not calculated run-time. Instead, using the method of the invention, once BLER has been calculated run-time (without using mapping tables) for the current MCS, a possible change of MCS is established by a simple comparison with two tabulated thresholds. That means to remit to the off-line simulations all the conceptual passages for the determination of the BLER thresholds. The off-line simulations have gone through the conceptual stages that will be detailed later on.

[0049] From above it can be appreciated that the method to perform link adaptation of the invention in subject is quite different from the prior art. In particular it is extremely simpler during run-time execution but also least complex during the off-line simulation devoted to prepare the tabulated thresholds. Obviously the run-time advantage pays the prize of the incapacity to catch the variability of the RF link, if not otherwise provided for; that because the variance is not calculated. The method of the invention provides an effective remedy to the outlined potentially drawback consisting of an additional set of tabulated thresholds for a channel with high variability. The additional set is particularly appreciable in presence of frequency hopping, that because its new thresholds of BLER better fit with the effects of an increased variability of the RF channel. The two sets of tabulated thresholds for channels with or without frequency hopping are both allocated to the BSS (PCU) and either enabled in the occurrence.

[0050] The approach taken for frequency hopping can be extended in line of principle to consider the possible practical RF scenarios. From simulation results it may be noticed that they can be grouped into a few significant cases. For instance, in a typical urban environment, only two different cases can be taken into account: a "low diversity" and a "high diversity" scenario. A first set of thresholds for the "low diversity" scenario should be selected if the cell is characterized by a low user mobility, such as: pico-cells, indoor cells, etc., without Frequency Hopping. A second set of thresholds for the "high diversity" scenario should be selected instead if the cell is characterized by a higher user mobility, such as: ≈50 Km/h mobile speed, or if Frequency Hopping is enabled. The method of the present invention provides the two set of thresholds for "high diversity" and "low diversity" RF channels, in that resolving the problem of the variability of the RF channel.

[0051] The impact of Incremental Redundancy (IR) with Link Adaptation (LA) needs some other considerations out of the mere variability of the RF channel. Some problems arising by combining IR with LA will be outlined in the following, then the solution of these problems by the method of the present invention will be introduced.

[0052] Incremental Redundancy together with link adaptation is known in the art. An exhaustive presentation of the problematic around IR with link adaptation is carried out in the International application number WO 00/49760, also filed by ERICSSON and completely taken into the standard ETSI **GSM 04.60.** A main problem solved by this secondly cited prior art is that of providing suitable overhead signalling messages to enable dynamic changing of the MCS during a connection, taking into account contrasting exigencies between Incremental Redundancy and pure Link Adaptation. A first type of said overhead messages is named LA/IR and corresponds to an additional bit inserted as a flag by the transmitting entity (i.e. the mobile station) in a control word of the RLC control blocks periodically transmitted in uplink to the receiving entity (the network). The LA/IR message provides an explicit request of the preferred operating mode, i.e. either link adaptation or incremental redundancy. This information can then be used by the network when selecting one of two predetermined rules for changing the MCS. For example, if the mobile station MS transmits the LA/IR field with a value which indicates that incremental redundancy is preferred, this implies that it currently has adequate memory capacity to continue to store blocks to support IR combining. This informs the network that the BTS can employ an MCS scheme more aggressive (less robust), taking the link quality estimate report into account. Alternatively, the LA/IR field may instead have a value which indicates that link adaptation is preferred by the mobile station. This may imply that the Mobile station lacks available memory and, therefore, cannot rely on incremental redundancy combining. When the network receives this message may then switch to a second MCS rule makes more conservative (more robust) MCS choices, based on the quality estimates, to ensure that the mobile station achieves sufficient performance without the incremental redundancy combining. Commands to change MCS are enclosed in downlink control blocks.

[0053] A second type of overhead messages of the two mentioned in WO 00/49760 is the value of an additional bit flag named RSEG/NRESEG by means of that the receiving entity informs the transmitting entity whether the MCS for retransmission should be the same or different than the MCS for new blocks transmissions. Before considering the reasons for sending RSEG/NRESEG message a general description of the MCS opportunities for EGPRS is needed. **TABLE 3 of APPENDIX 1** shows that the EGPRS MCS are divided into different families named A ($A_{padding}$), B and C. Each family has a different basic unit of payload: 37 (and 34), 28 and 22 octets respectively. Different code rates within a family are achieved by transmitting a different number of payload units within one Radio Block. For families A and B, 1, 2 or 4 payload units are transmitted, for family C, only 1 or 2 payload units are transmitted. When 4 payload units are transmitted (MCS-7, MCS-8 and MCS-9), these are split into two separate RLC blocks (i.e. with separate sequence numbers and BCSs) within the same Radio Block. These blocks in turn are interleaved over two bursts only, for MCS-8 and MCS-9. For MCS-7, these blocks are interleaved over four bursts. All the other MCSs carry one RLC block interleaved over four bursts. When switching to MCS-3 or MCS-6 from MCS-8, 3 or 6, padding octets, respectively, are added to the data octets. The highlighted structure of the MCSs schemes offers more than one retransmission opportunity to cope with change in the RF channel, for example it's possible under certain restriction, that the message originally pertaining one radio block be retransmitted with more, or less, robust MCS scheme. A change of MCS for the retransmitted message involving a splitting of the payload is said re-segmentation. In case the receiving entity were the network, the downlink control blocks transporting a suitable message include an MCS command which tells the mobile station which MCS should be used for transmitting uplink RLC blocks. The RSEG/NRESEG bit can also be added to the downlink control blocks. In this context a NRSEG asserted (re-segment bit = 0) can be interpreted by the mobile station as meaning retransmissions by the mobile station using the same MCSs as the initial transmissions of RLC blocks; on the other hand a NRSEG negated (re-segment bit = 1) should be interpreted by the mobile station as meaning that blocks to be retransmitted could be re-segmented and transmitted using different MCSs than the initial one. In the latter case, the specific MCS to use for retransmission can be determined by a predetermined rule stored in the receiving entity (mobile station).

[0054] A help in retransmission come from ETSI **GSM 04.60,** paragraph titled "Acknowledged mode operation - Additional functionality in acknowledged EGPRS TBF Mode", in which a procedure is proposed which allows the receiver to operate either in type I or type II hybrid ARQ mode. This procedure says that according to the link quality, an initial MCS is selected for an RLC block. For the retransmissions, the same or another MCS from the same family of MCSs can be selected. E.g. if MCS-7 is selected for the first transmission of an RLC block, any MCS of the family B can be used for the retransmissions. Further, RLC data blocks initially transmitted with MCS4, MCS-5, MCS-6, MCS-7, MCS-8 or MCS-9, can optionally be retransmitted with MCS-1, MCS-2 and MCS-3 respectively, using two radio blocks. In this case, the SPlit Block indicator (SPB) in the header shall be set to indicate that the RLC data block is split, and the order of the two parts. For blocks initially transmitted with MCS-8 which are retransmitted using MCS-6 or MCS-3, padding of the first six octets in the data field shall be applied, and the Coding and Puncturing Scheme (CPS) field shall be set to indicate that this has been done. However, if the transmitter side is the MS and the re-segment bit is not set, the mobile station shall use an MCS within the same family as the initial MCS without splitting the payload for retransmission. The RLC data blocks shall first be sent with one of the initial code rates (i.e., the rate 1/3 encoded data is punctured with the Puncturing Scheme (PS) 1 of the selected MCS). If the RLC Data Block has to be retransmitted, additional coded bits (i.e., the output of the rate 1/3 encoded data which is punctured with PS 2 of the prevailing MCS) shall be sent. If all the codewords (different punctured versions of the encoded data block) have been sent, the procedure shall start over and the first codeword (which is punctured with PS 1) shall be sent followed by PS 2 etc. RLC data blocks which are retransmitted using a new MCS shall at the first transmission after the MCS switch be sent with the puncturing scheme indicated in the **APPENDIX 1 - TABLE 4.** Furthermore, it is mandatory for an EGPRS MS receiver to be able to perform joint decoding among blocks with different MCS's if the combination of MCS's is one of the following:

- MCS-5 and MCS-7,
- MCS-6 and MCS-9.

[0055] The long explanation of the LA/IR technique has twofold meaning, a first one attends to clarify enough this complex argument at the advantage of the disclosure, a second one is that to highlight the lack of indication in the prior art useful to understand the influence of the IR mechanism on the decisional thresholds of BLER. The only reasonable inference on LA/IR from the teaching of the prior art is that IR take over pure LA in case of retransmission with infinite memory pad, but considering the more realistic case of memory saturation, LA is also activated to avoid frequent retransmission. There is a sort of pronounced antagonism between LA and IR at the lower C/I, and the higher BLER values are involved consequently. As the invention in subject is based on the recurs to particular BLER thresholds to maximize the throughput, an important question is that to take realistic thresholds in presence of IR. The prediction of LA/IR interactions is not an easy task at all, because, beyond the probabilistic nature of the phenomenon, the knowledge

of the precise memory size is also required. Memory size at the mobile station side depends on the customer preferences about costs and dimensions of the apparatuses and can't be planned by BSS producer consequently.

[0056] The present invention solves the outlined technical problem starting from the introduction of a variable IR_status which gives continuously updated information to the receiving entity (either the network or the mobile station) about the efficiency of Incremental Redundancy, as disclosed in a relative dependent claim. The evaluation of IR_status is quite simple. Filtered values assumed by the variable IR_status are taken to update the BLER thresholds consequently. Updating is performed by a linear interpolation between two extreme conditions, namely: BLER thresholds relative to lack of IR and BLER thresholds relative to perfect IR. Intermediate and more realistic conditions, so as the two extreme ones, are automatically managed through the updated threshold mechanism. The outlined contrasting behavior between LA and IR, since now remarkable source of problems in the determination of the best adaptation strategy, is no more a problem with the method of the invention extended to the Incremental Redundancy.

[0057] Last argument of the invention is a modified Power Control algorithm having a different goal then the traditional one. The modified algorithm attempts to maintain a C/I_{target} target value for the duration of the whole TBF. The C/I_{target} target is associated to a Peak Throughput per timeslot decided as "Target performance". The association is performed through a curve that represents the maximum achievable Throughput versus C/I. This curve belongs to those simulated off-line during the preliminary step of the Link Adaptation subject of the present invention, in particular to a set having care of the incremental redundancy. Although the upper goal of Power Control, Link Adaptation continues to adapt to radio conditions, switching from one MCS to another, in order to optimize performance on net throughput. This may happen due to the fact that the power control cannot be "perfect" and therefore the actual C/I ratio may be different from the target one. From above it can be argued that the Modified Power Control algorithm complete the Link Adaptation of the present invention working in synergy with it; in that resolving the outlined controversy of the traditional Power Control. Besides, contrarily to the power control of the first cited document of the prior art, it need not separate optimization for each available MCSs.

[0058] From all the above considerations the following substantial advantages of the proposed invention emerge, namely:

- link adaptation runs independently on quality measures, however performed on the ongoing RF signal for traditional Power Control and Handover procedures;
- the variability of the RF channel is neutralized in advance in the adaptation;
- the memory size for Incremental Redundancy is managed in a transparent way;
- power control pursues same goal as link adaptation.

BRIEF DESCRIPTION OF THE DRAWINGS

[0059] Further objects and advantages of the present invention will be made clear by the following detailed description of an embodiment thereof and the annexed drawings given for purely non-limiting explanatory purposes and wherein:

- **fig. 1** shows an GSM and GPRS radiomobile network operating in conformity with the method of the present invention;
- **fig. 2** shows a Time Division Multiple Access (TDMA) multiframe structure common to the GSM and GPRS of fig.1;
- **fig. 3a** shows a GSM traffic channel multiframe;
- **fig. 3b** shows an GPRS traffic channel multiframe;
- **fig. 4** shows the mapping of higher level frames into radio blocks belonging to the EGPRS multiframe of fig.3b;
- **fig. 5** shows the functional blocks of a mobile station MS/UE of fig.1 operating in conformity with the method of the present invention;
- **fig.6** shows the functional blocks of a base station BTS of fig.1 operating in conformity with the method of the present invention;
- **figures 7, 8, 9, 10, 11, 12, 13, 14** show graphic representations of simulation results used to implement the method of the present invention;
- **figures 15 and 16** show respective flow charts of the link adaptation method of the present invention;
- **fig.17** shows a graphic representation used to implement the power control function at the Um interface of fig.1.
- **APPENDIX 1 - TABLE 1** includes coding parameters for GPRS coding schemes.
- **APPENDIX 1 - TABLE 2** includes coding parameters for EGPRS modulation and coding schemes.
- **APPENDIX 1 - TABLE 3** represents payload families used in the EGPRS coding schemes.
- **APPENDIX 1 - TABLE 4** includes Puncturing Schemes for EGPRS.
- **APPENDIX 1 - TABLE 5** includes modulation and coding schemes to be used for retransmissions when re-segmentation is not enabled.
- **APPENDIX 1 - TABLE 6** includes modulation and coding schemes to be used for retransmissions when re-seg-

mentation is enabled.

DETAILED DESCRIPTION OF AN EMBODIMENT OF THE INVENTION

[0060] The arguments of **Figures 1, 2, 3a, 3b and 4,** so as **TABLES 1, 2, 3, and 4** of **APPENDIX 1** have already been duly discussed above in the text.

[0061] **Figure 5** shows a block diagram of a mobile station MS/UE suitable to implement the present invention in conjunction with the BSS subsystem of fig.1. The mobile station MS/UE includes a Transmitting Section and a Receiving Section both controlled through a Control Processor that further controls a Frequency Synthesizer & Hopping unit common to the two sections. A Duplexer filter conveys to the antenna the RF output signals of the Transmitting section and to the input of the Receiving section the RF signal received on the antenna. For the sake of simplicity an oscillator and a TDMA timing generator are not shown in fig.5. The Transmitting Section includes the following functional blocks: Input devices, Speech coder, Channel coder, Interleaver, Ciphering, Burst formatter, GMSK / 8-PSK Modulator, BB/IF/RF UP converter, and RF Power amplifier. Input devices include a microphone with relative A/D converter and a Keyboard & Adapter. The Receiving Section in its turn includes the following functional blocks: Image filter, RF amplifier, RF/IF/BB Down converter, LEV, CH filter, A/D converter, Correlator and MLSE (Viterbi) estimator, Burst disassembler, Deciphering, De-Interleaver, Channel decoder, Speech decoder & Voice amplifier, Output devices (Earphone, PC Monitor, Fixed Disk, etc.). In conformity with its A, B, or C operative class, the mobile station MS/UE is able to operate with both voice and data input devices, simultaneously or not. Class A users have one time slot allocated for speech and one ore more others to the EGPRS service. Dual considerations apply to the Output devices. As already mentioned the present invention is prevalently addressed to packet data, so the blocks Input devices and Output devices will exemplify known data terminals for inputting or outputting data respectively. Those terminals include pads and adapter circuits for synchronizing, storing, adapting format and rate of the incoming/outgoing digital blocks. Considering the Transmitting section at first, Channel coder accepts data from Input devices and provides a relevant EGPRS coding scheme, selected from those reproduced on TABLES 1 and 2. For this aim a CPS-TX-SEL signal is outputted from the Control Processor. Channel coder provides for: block code, parity code, convolutional code and fire code; it further accepts and codifies DATA-INS signalling RLC blocks (such as measures) from the Control Processor. Coded blocks are sent to the cascade of Interleaver, Ciphering, and Burst formatter to perform the relative digital treatments as explained in the introduction. A formatted burst is delivered to the GSM / 8PSK Modulator which starts performing a differential encoding followed by either a GMSK or 8-PSK modulation. Control Processor selects the modulation type by sending to the Modulator a MOD-TX-SEL signal, always in respect of the MCS schemes listed in TABLE 2. The base band analog modulated signal is firstly translated to IF frequency and then to RF frequency by means of suitable up conversion mixers; each conversion stage is followed by a band pass filtering stage. The RF transmission signal reaches the input of a variable gain Power amplifier whose output is coupled to a transmission port of a Duplexer filter coupled to the Mobile station antenna. The downlink RF signal coming from the BTS reaches the Mobile station antenna and leaves a receiving port of the Duplexer filter, crosses an Image filter and reaches the input of a Reception low noise amplifier whose output is connected to a frequency down converter. The down conversion is carried out by two cascaded stages: a first one converts from RF to IF, and the second one from IF to base band BB. The second stage also splits the converted signal into the in-phase I and in-quadrature Q components. The base band I, Q components are filtered by two channel filters CH matched to the transmitted pulse and then analog-to-digital converted. The two copies of the digitalized reception burst arrive at the two inputs of a Correlator/Synchronizer, acting like a matched filter to the training sequence, which extracts the correlation peak for detecting the initial instant of the transmission. The same correlative process also estimates the pulse response of the channel supplied to an MLSE estimator based on the Viterbi algorithm. This algorithm acts on a sequentially built-up trellis having as many nodes (reiterated at each symbol time T) as the states $S = M^L$ of the receiver, corresponding to all the possible combinations generated from M words (symbols)) of a modulation alphabet over L symbol times (where L is the significant length of the initially estimated channel pulse response). Starting from a known initial state, the progressive path along the trellis will depend on the effective transmitted sequence. All the possible transmitted sequences are distinguished each other through a respective path metric which constitutes the Likelihood function to be gradually maximized by accumulating transition metrics. At every new symbol time M transition metrics Δ are calculated in correspondence of the M branches departing from each preceding node to reach a number M of successive nodes. A transition metric (or branch metric) is the Euclidean distance between the level of the received symbol and the level that should have been received in correspondence of a supposed transition on the trellis. Among all the branches departing from a node only a survivor one is selected to prolong a trellis path passing through that node, namely that having the maximum actual path metric. So doing, a drastic cut of the complexity is performed because the original number of states is maintained at each step. Among all the survived paths at the time T the candidate sequence is the one which has the maximum path metric. Going back along the trellis for a certain number of steps it can be appreciate that only a path survives, which is associated to a segment of the transmitted sequence. More precision is obtained delaying the decision of the MLSE estimator until

the end of the burst. At the output of the MLSE estimator a copy of the original burst is reproduced and each bit is accompanied with three bits soft decisions indicating its received level. The estimated burst is delivered to a cascade of the following blocks: Burst disassembler, Deciphering, De-interleaver, and Channel decoder; the last carries out the specified operations in respect of TABLES 1 and 2 by exploiting soft decisions. Control Processor generates the following two signals: MOD-RX-SEL and CPS-RX-SEL towards MLSE estimator and Channel decoder respectively. That because modulation and/or code scheme of the received signal can differ from the transmitted ones. MLSE estimator operates with either GMSK or 8-PSK modulation, obviously with different trellis and branch metric expression. Channel decoder uses Soft decisions to carry out convolutional decoding and also takes advantage from the mentioned Incremental redundancy strategy supported by an Incremental Redundancy buffer for temporarily storing RLC blocks to be retransmitted under ARQ. A buffer overflow activates a signal IRout directed to the Control Processor. Decoded RLC signalling blocks, indicated with DATA-EXTR, are extracted and sent to the Control Processor for the correct interpretation and execution (such as: Power control, Timing Advance, Handover, etc.). Channel decoder detects and counts errors before error correction and informs the Control Processor by sending a signal BER having the usual meaning of Bit Error Rate. Since decoding is good an OK Flag is set. Decoded RLC blocks concerning traffic are sent to the appropriate output devices in conformity with the selected A, B or C user class. Control Processor governs the main operational procedures of the Mobile station MS/UE through a first and a second group of signals indicated as TRANSMITTING SECTION control and RECEIVING SECTION control, respectively directed to the two sections. Among these signals the following three are pointed out: MAIO, RSSI and PC. MAIO is directed to the Frequency Synthesizer & Hopping unit in order to provide indication for frequency hopping and handover. Signal RSSI is generated from a circuit LEV which samples, A/D converts, and measures the strength of the received signal, and noise during idle. Control Processor block includes a memory RAM for temporarily store Level 2 and Level 3 signalling messages.

[0062] **Figure 6** shows a block diagram of a Base Transceiver Station (BTS) suitable to implement the present invention in conjunction with the BSS subsystem of fig.1 and the Mobile Stations MS/UE of fig.5. The mobile BTS includes a Transmitting Section and a Receiving Section both controlled through a BTS Control Processor that further controls a Frequency Synthesizer & Hopping unit common to the two sections. The two Sections and the BTS Control Processor are connected to an A-Bis INTERFACE functional block for receiving/outputting one or more PCM link at 2 Mb/s or PCU frames incoming from or outgoing to the BSC (fig.1). A Duplexer filter conveys to the antenna the RF output signals of the Transmitting section and to the input of the Receiving section the RF signal received on the antenna. For the sake of simplicity a clock generator/extractor and a TDMA timing generator are not shown in fig.6. The Transmitting Section includes the following functional blocks: Base band processing 1...n, GMSK or 8-PSK digital modulators 1....n, MULTICARRIER DIGITAL TRANSMITTER. The Receiving Section includes the following functional blocks: Base band processor 1,...,n, Equalizer & Demodulator 1,....,n, MULTICARRIER DIGITAL RECEIVER, and an Image filter. Starting from the Transmitting Section, the A-bis INTERFACE block extracts from the PCM link or PCU frames all the n elementary fluxes concerning CH1....CHn channels relevant to the n users. CH1....CHn fluxes reach respective Base band processors to undergo all the digital treatments as: coding (parity, convolutional fire), interleaving, ciphering, burst formatting, and differential coding. Convolutional coding provides a relevant EGPRS coding scheme, selected from those reproduced on TABLES 1 and 2. The n coded signals outputted from the Base band processors reach as many GMSK/8-PSK digital modulators to be digitally modulated in respect of the MCS schemes listed in TABLE 2. The n modulated digital signals reach as many DUCs (Digital Up Converters) inside the MULTICARRIER DIGITAL TRANSMITTER. Each DUC further receives a respective local oscillator signal $f_{IF\text{-}DUC}$ for the translation of its base band input signal to a prefixed position inside the overall Intermediate Frequency band. For this aim the $f_{IF\text{-}DUC}$ signals are digital sinusoids. The n IF digital signals are summed up by a digital adder working at the higher $f_{IF\text{-}DUC}$ frequency, and the multicarrier IF resulting signal is D/A converted and wide band filtered before reaching the input of an IF/RF mixer piloted by a $f_{OL\text{-}TX}$ local oscillator signal to the up conversion at radiofrequency. The RF signal at the output of the mixer is sent to an RF power amplifier. The output of the RF power amplifier is connected to the TX port of the Duplexer filter, while the RX port is connected to the Image filter placed at the input of the MULTICARRIER DIGITAL RECEIVER. The RF filtered signal is amplified and down converted to IF by an RF/IF mixer piloted by a $f_{OL\text{-}RX}$ local oscillator signal. The multicarrier analog IF signal is anti-alias filtered and fed to the input of n DDCs (Digital Down Converters) inside the MULTICARRIER DIGITAL RECEIVER. Each DDC further receives a respective local oscillator signal $f_{IF\text{-}DDC}$ for the translation to base band its input signal relevant to a prefixed position inside the overall Intermediate Frequency band. For this aim the $f_{IF\text{-}DDC}$ signals are digital sinusoids. The n digital base band signals reach as many Equalizer & Demodulator to be demodulated in respect of the MCS schemes listed in TABLE 2. The same arguments as the Viterbi's estimator of fig.5 are still valid. The demodulated signals are sent to the Base band processors to undergoes: Burst disassembling, Deciphering, De-interleaving, and Channel decoding in respect of TABLES 1 and 2. Finally the decoded data relevant to CH1,...CHn channels are delivered to the A-bis INTERFACE functional block to be assembled into one 2 M/bit outgoing PCM link or PCU frames. Control Processor block includes a memory RAM for temporarily store Level 2 and Level 3 signalling messages for all the n users. The BTS Control Processor governs the main operational procedures of the Mobile station MS/UE through a first and group of signals indicated as "TRANSMITTING

SECTION control", "Signalling insertion"; and a second group of signals indicated as "RECEIVING SECTION control", "Signalling extraction". Among these signals a MAIO group is directed to the Frequency Synthesizer & Hopping unit in order to provide indication for frequency hopping and handover as far as concerns all the DUC and DDC circuits. Control Processor block includes a memory RAM for temporarily store Level 2 and Level 3 signalling messages for all the n users. Extracted signalling concerns, for example: measures transmitted uplink by all the Mobile stations (level, BER, C/I, OK flag, etc.), the statuses of the IRout overflow indicators, etc. Inserted signalling concerns, for example: Power control commands directed to each Base band processor, Timing advance commands, selection of the individual MCS scheme for transmission and/or reception, etc.

[0063] The reference frame of a known GSM-EGPRS system has been completed at this point of the disclosure. So the basis for the introduction of the features typical of the invention are given. The relevant means of the invention to carry out uplink and/or downlink TBF link adaptation constitute a particular combination of known and new means like the following list, in which when not expressly mentioned they are preferably allocated to the PCU and either confined in the firmware or in dedicated circuits:

- memory matrix tables for memorizing as many sets of digital values intended as BLER thresholds; the tables being managed by the Packet Control Unit (PCU). The thresholds being calculated off-line in a way that will be soon illustrate and they are valid both for uplink and downlink adaptation;
- means of the PCU for the selection of the tables;
- means allocated both to the BTS and the mobile stations for decoding RLC received blocks, optionally capable of joint decoding Incremental Redundancy bits;
- means allocated both to the BTS and the mobile stations for detecting and storing RLC blocks erroneously received;
- means allocated both to the BTS and the mobile stations for retransmitting erroneously received blocks;
- means for calculating BLER of an active TBF by filtering a variable indicating the RLC blocks not correctly received;
- means for checking the performance of the Incremental Redundancy detection;
- means for filtering a variable indicating the effectiveness of the Incremental Redundancy detection;
- means for continuously updating the BLER thresholds on the basis of said effectiveness variable;
- means to compare the calculated BLER with the updated BLER thresholds in order to obtain a criterion for changing the actual MCS;
- means of the PCU to command a new MCS on the basis of said criterion for changing the actual MCS;
- means of the BSC for updating the transmission power level of each uplink/downlink channel in order to maintain a fixed target throughput independently on the MCSs.

[0064] With reference to the **Figures 7 to 14** the preliminary off-line simulation step useful for determining the various sets of BLER thresholds is now considered. Those Figures are to be considered two at a time, such as: Fig.7 and 8; 9 and 10; 11 and 12; 13 and 14. The arguments relative to the first couple of Figures 7 and 8 are generally still valid for the other couple of figures. **Fig. 7** shows some curves of net throughput (kbit/s) in function of C/I (dB) for several Modulation and Coding Schemes. **Fig. 8** shows correspondent curves of BLER (dB or %) in function of C/I (dB) for the same MCSs of fig.7. Four MCSs are represented in fig.7 indicated with a, b, c, d; they respectively coincide with MCS1, MCS3, MCS6, and MCS9 of **TABLE 2.** It can be appreciate that the listed MCSs is a subset of all the possible MCSs constituting a sequence of MCSs arranged by increasing nominal throughputs. Curves of **fig.7** are referred to a standard channel TU3 (Typical Urban - 3 ray model) without Frequency Hopping and without Incremental Redundancy (only Type I ARQ is admitted), they are valid for both uplink and downlink TBFs. The depicted values are the result of a computer simulation refined and validated through on field measures. Curves of **fig.7** are derived from curves of **fig. 8** by using the relation (1).

[0065] Because of the trends of the various MCS curves of fig.7 are not similar to that of parallel lines, six different cross-points are visible in correspondence of as many values of C/I. Cross-points are characterized by equal net throughputs for at least two MCS curves. Cross-points relevant for the present invention are only the three relative to adjacent MCSs in the ordered sequence, namely: a-b, b-c, and c-d. In order to maximize throughput the higher order MCS should be selected at the right of the switching point, while the lower order MCS should be chosen when the RF channel conditions are at the left of the cross point. This behavior is due to the decreasing protection of the higher MCS at the lower C/I and the consequent retransmission of the errored radio blocks. Referring to the previous cross-points of fig.7 the 'ideal' switching points between two adjacent MCSi could be the following:

MCS a ↔ MCS b:	C/I ≈ 1.5 dB
MCS b ↔ MCS c:	C/I ≈ 7.5 dB
MCS c ↔ MCS d:	C/I ≈ 16 dB

[0066] But C/I values are difficult to estimate in a real network, while other parameters, such as BLER, can be calculated directly. The Link Adaptation algorithm here proposed will then be based on direct BLER measurements. The previous calculated 'ideal' C/I switching points now correspond to the following 'ideal' couples of BLER thresholds mapped on the curves of **fig.8:**

$$\text{MCS a} \leftrightarrow \text{MCS b: C/I} = 1.5 \text{ dB} \Rightarrow \text{BLER}_{\text{MCS1}\rightarrow\text{MCS3}} = \text{Tab, BLER}_{\text{MCS3}\rightarrow\text{MCS1}} = \text{Tba;}$$

$$\text{MCS b} \leftrightarrow \text{MCS c: C/I} = 7.5 \text{ dB} \Rightarrow \text{BLER}_{\text{MCS3}\rightarrow\text{MCS6}} = \text{Tbc, BLER}_{\text{MCS6}\rightarrow\text{MCS3}} = \text{Tcb;}$$

$$\text{MCS c} \leftrightarrow \text{MCS d: C/I} = 16 \text{ dB} \Rightarrow \text{BLER}_{\text{MCS6}\rightarrow\text{MCS9}} = \text{Tcd, BLER}_{\text{MCS9}\rightarrow\text{MCS6}} = \text{Tdc.}$$

[0067] Net throughput is then maximized changing the MCS according to these BLER threshold values. If actual BLER falls below the upgrade threshold (Tab, Tbc, Tcd) the algorithm switches to the next (less protected) available MCS. If actual BLER instead exceeds the downgrade threshold (Tab, Tbc, Tcd) the algorithm switches to the previous (more protected) available MCS. For example, if BLER goes below Tbc, while using MCS b, then a change to MCS c will be decided. On the contrary, if BLER goes above Tba, while using MCS b, then a change to MCS a will be decided.

[0068] If the RF environment changes, the MCS's performances curves change as well. Therefore the 'ideal' switching points depend on the actual RF environment. As an example, 'ideal' switching points may be different if Frequency Hopping is enabled or disabled in the network. Though the possible RF scenarios are virtually infinite, as already anticipated in the introduction, in a typical urban environment, only two different cases can be taken into account: a "low diversity" and a "high diversity" scenario.

[0069] The "low diversity" scenario corresponds to the family of curves represented in **Figures 7 and 8** and should be selected if the cell is characterized by a low user mobility, such as: pico-cells, indoor cells, etc. without Frequency Hopping.

[0070] The "high diversity" scenario corresponds to the family of curves represented in **Figures 9 and 10** and should be selected if the cell is characterized by a higher user mobility, such as ≈50 Km/h mobile speed, or if Frequency Hopping is enabled. Simulation results represented in Figs. 9 and 10 have been obtained in absence of IR.

[0071] For each specific RF scenario different upgrade switching points and downgrade switching points are derived through simulations and on field measures. These values of the switching points constitute as many sets of thresholds stored in matrix tables. Once the particular RF scenario has been assigned, the corresponding matrix table is selected, containing all the ideal switching points (downgrade/upgrade switching points from/to all MCSs) for that case. The initial MCS has to be defined as said later on:

[0072] Things are further complicated when type II Hybrid ARQ (Incremental Redundancy) is utilized. In the Figures 11, 12 and 13, 14 simulation results with IR (and infinite memory) are presented for the same scenarios described above. More precisely, simulation results represented in **Figures 11 and 12** concern cells characterized by "low diversity" in presence of Incremental Redundancy. In this case it can be seen that MCS d outperforms all others MCS for a wide range of C/I ratios and the setting of the switching points will require some further considerations. Simulation results represented in **Figures 13 and 14** concern cells characterized by "high diversity" in presence of Incremental Redundancy. Even here further considerations are necessaries. In any case it should be noticed that again, even in presence of Incremental Redundancy, the resulting performance depends on the actual RF scenario. Moreover results depend on the amount of memory available for Incremental Redundancy. Anyway, as a result, when IR is taken into account, different BLER threshold values should be considered. Even these values should be stored in matrix tables, one for each possible RF scenario.

[0073] With reference to the **Figures 15 and 16,** the Link Adaptation method subject of the present invention is discussed. For the sake of simplicity the method is like a flow-chart of a program which controls a microprocessor inside the PCU (fig.1). In the reality the various steps of the program interact with the involved protocol procedures and signalling. The previous off-line step for obtaining the BLER threshold matrix tables shall be considered as a preliminary part of the method. **Fig.15** concerns a simplified method valid for a packed data scenario without Incremental Redundancy and either characterized by low or high variability. **Fig.16** differs from fig.15 in that Incremental Redundancy is considered. Matrix tables relative to the **Figures 8, 10, 12,** and **14** have been respectively indicated as **Table A, B, C,** and **D.**

[0074] The method of **fig.15** starts with step **S1** which addresses the TBFs adaptation either uplink or downlink. Presently uplink TBFs are considered, successively the modifications for downlink TBFs will be introduced. In the subsequent step **S2** the connection is established and the Initial Modulation and Coding Scheme is decided. The initial MCS will be set by default, unless some other information is available. In step **S3** at the network side value of BLER

is continuously updated, at each received radio block, by checking if RLC blocks have been carefully received or not. BLER at instant n, for a given TBF connection, is obtained by a digital filter having a pulse response exponentially decreasing with time discrete n as indicated by the following law:

$$BLER_n = f_1(BLER_{n-1})+f_2(s_n) \tag{2u}$$

where:

- n is the iteration index spanning one radio block period of 20 ms;
- $s_n = 0$ if the RLC block at instant n has been correctly received (and the MCS is the "commanded MCS");
- $s_n = 1$ if the RLC block at instant n has not been correctly received;
- $$s_n = \frac{1}{K}\sum_{k=1}^{K} s_{n,k} \tag{3u}$$

  if more than one RLC block is received s_n is the average of the values calculated for single blocks. De facto more than one RLC block for a given TBF can be received at the same time instant n, due to 1) multislot allocation, 2) MCSs supporting two RLC blocks at a time.
- $f_1(BLER_{n-1})$ is a first weight function of the preceding filtered BLER value relative to the "commanded MCS" (i.e. actual MCS) blocks only, taking values inside the interval 0 - 1;
- $f_2(s_n)$ is a second weight function of the variable s_n, taking values inside the interval 0 - 1;

[0075] Taking into consideration the teaching of standard ETSI **GSM 05.08** about time filtering of quality variables, for analogy, expression (2u) now assumes the following expression:

$$BLER_n = (1-\beta\cdot\frac{x_n}{R_n})\cdot BLER_{n-1}+\beta\cdot\frac{x_n}{R_n}\cdot s_n \tag{2u'}$$

where:

- n is the iteration index spanning one radio block period of 20 ms;
- x_n is equal to 1 if "at least" one RLC block for the considered TBF with the "commanded MCS" is received at time instant n, otherwise is set to 0;
- s_n has been already defined;
- β is the forgetting factor:

$$\beta = 1/T_{AVG},$$

[0076] T_{AVG} being the filtering period in multiples of a radio block;

- R_n denotes the reliability of the n_{th} BLER measurement and is expressed as follows:

$$R_n = (1-\beta)\cdot R_{n-1}+\beta\cdot x_n; \qquad R_{-1} = 0 \tag{4u}$$

  R_n is the output of a running average filter that helps to keep track of the reliability of the filtered BLER measurements. In fact R_n is used in (2u') to decide the weight between the new measurement (s_n) and the old measurements ($BLER_{n-1}$). Looking at the formulas, it comes out that at time instants where no measurement exists (no RLC blocks are received for the considered TBF), $BLER_n$ will not be updated. On the contrary, when a measurement exists, $BLER_n$ will be updated weighting new and old contributions, so to obtain the desired exponentially decreasing (with discrete time n) filter impulse response. The reliability filter is initialized at the beginning of a transmission (n=0) setting $R_{-1} = 0$.

By the comparison of expression (2u) with (2u') it results that:

$$f_1(BLER_{n-1}) = (1-\beta\cdot\frac{X_n}{R_n})\cdot BLER_{n-1}$$

$$f_2(S_n) = \beta\cdot\frac{X_n}{R_n}\cdot s_n.$$

Besides the two weight functions $f_1(BLER_{n-1})$ and $f_2(s_n)$ have balanced weights, so that an arbitrary weight increasing of $f_2(s_n)$ also involves an equal weight decreasing of $f_1(BLER_{n-1})$, and vice versa.

[0077] In next step **S4** the presence of Frequency Hopping is checked. If the answer in step **S4** is negative, the case of low diversity environment is checked in the successive step **S5.** Affirmative answer in step **S4** enters step **S6** in which the BLER filtered at step **S3** is compared to the upgrade and downgrade thresholds stored in **Table A.** Negative answer in step **S4** enters step **S6'** where the filtered BLER is compared to the thresholds stored in **Table B.** The comparison using **Table B** is also performed if Frequency Hopping were found active in the preceding step **S4.** Thresholds could be generalized in this way: put MCSx the actual MCS, MCSy the next available less protected one, and MCSz the previous available more protected one, then the appropriate thresholds will be:

Upgrade thresholds (UP_th_n): $BLER_{MCSx\rightarrow MCSy}$

Downgrade thresholds (DN_th_n): $BLER_{MCSx\rightarrow MCSz}$.

[0078] Reaching the step **S7** either from **S6** or **S6',** the occurrence is checked of an MCS switching in consequence of the previous comparisons. If in step **S7** the actual value of BLER doesn't cross any thresholds the subsequent step **S8** performs an unitary increment of index n, then in step **S9** the active state of the actual TBF is monitored. Until TBF is active the respective BLER is continuously monitored from the cycle of steps **S3 - S10** to check the conditions for switching from the actual MCS; if during the cycle the TBF elapses the incoming step **S10** resets BLER and R and the program waits for another TBF. If during the cycle **S3 - S10** the actual BLER falls below the value UP_th_n, then MCSx is switched to MCSy in step **S11.** Alternatively, if during the cycle **S3 - S10** the actual BLER exceeds the value DN_th_n, MCSx is switched to MCSz in step **S11.** When commanding the new MCS to the MS, in a PACKET UPLINK ACK/NACK or PACKET TIMESLOT RECONFIGURE message, the PCU can also set the re-segment bit to the proper value. In general, for retransmissions, setting the re-segment bit to '1' requires the mobile station MS/UE to use an MCS within the same family as the initial transmission and the payload may be split. Instead setting the re-segment bit to '0' requires the mobile station shall use an MCS within the same family as the initial transmission without splitting the payload. **TABLES 5 and 6** of **APPENDIX 1** show MCS schemes to use for retransmission after switching to a different MCS. **TABLE 5** is valid for re-segment bit = 1, while **TABLE 6 is valid** for re-segment bit = '0'. According to the invention, in the case under description (no Incremental Redundancy mode), the re-segment bit is always set to "1". Whenever the Modulation and Coding Scheme is changed, BLER and R variable are set to zero in the successive step **S12** and the filtering process is re-started from step **S3.**

[0079] Additional advantages of the disclosed method are mostly due to the filtering step **S3,** they are:

- Considering that at each iteration index n used in digital filter (2u) and (2u') (20 ms) could not correspond an RLC block for the intended TBF, due to the MAC scheduling mechanism, and that, on the contrary, a constant BLER filtering window is preferable in expression (2u'), then the reliability filter (4u) provides the way to keep constant the "actual" BLER filtering window, in that independent on the number of TBFs multiplexed on the same TS. Consequently the BLER digital filter (2u') is taken back to the RLC blocks effectively received in order to maintain the right exponentially decreasing impulse response.
- Only blocks encoded with the present MCS contribute to BLER calculation. In other words, retransmissions with a different MCS don't have any impact on BLER calculation for the actual MCS.

[0080] With reference to the **fig.16** changes in respect to the **fig.15** are now discussed to be introduced in the link adaptation method due to the Incremental Redundancy. The first five steps **S1** to **S5** are the same as those of fig.15, in particular the filtering step **S3.** Additional problems arise when the actual thresholds for the BLER comparison shall be determined. These problems are of different nature and must be checked consequently, so step **S6 (S6')** is delib-

erately introduced for this aim. During this step the following routine is executed to set the logic value of a variable IR_check that contributes to the issue of an IR_status variable which gives information about the efficiency of Incremental Redundancy at the BTS:

IF

{

there has been an header error (this implies that IR for the expected block(s) is useless),

OR

if memory for IR is exhausted (no IR is possible for the expected block(s)),

OR

if soft decisions could not have been stored due to any other reason (again, no IR for the expected block(s))

}

THEN IR at time instant n is considered as "not working",

$IR\_check_n=0$

ELSE IR at time instant n is considered as "working",

$IR\_check_n=1$.

[0081] The IR_status is then filtered in step **S7 (S7')** using the same approach used for BLER in step **S3;** in particular using a digital filter having a response exponentially decreasing with discrete time n as indicated by the following law:

$$IR\_status_n = f_1(IR\_status_{n-1})+f_2(IR\_check_n) \qquad (5u)$$

were function f_1 and f_2 follow the same laws as used in the BLER calculation. The analogy is extended to the most detailed function:

$$IR\_status_n = (1-\beta\cdot\frac{X_n}{R_n})\cdot IR\_status_{n-1}+\beta\cdot\frac{X_n}{R_n}\cdot IR\_check_n \qquad (6u)$$

where: x_n, R_n, and β are the same values used in the BLER calculation.

[0082] Differently from the preceding method of fig.15, BLER thresholds stored in the matrix tables are not immediately usable, since such thresholds depend on the used MCS but on the IR efficiency as well. So the successive step **S8(S8')** is charged to calculate suitable thresholds for taking IR into account. The new thresholds are the result of a linear interpolation between two extreme cases, namely: perfect IR (IR_status = 1), and IR totally lacking (IR_status = 0). Each case making reference to its own matrix tables. Absence of IR needs tables **A** and **B,** while perfect IR needs tables **C** and **D,** besides tables **A** and **C** simulate low diversity channels respectively without and with IR, while tables **B** and **D** high diversity channels respectively without and with IR. Consequently the linear interpolation in step **S8** taking care of low diversity channels recurs to tables **A** and **C,** while the linear interpolation in step **S8'** taking care of high diversity channels recurs to tables **B** and **D.**

[0083] Indicating with $BLER_{MCSx\_wIR\rightarrow MCSy\_wIR}$, and $BLER_{MCSx\_wIR\rightarrow MCSz\_wIR}$ respectively the new upgrade UP_th_n and downgrade DN_th_n thresholds for perfect IR, the linear interpolations calculated either in step **S8** or **S8'** assume the following expressions:

$$UP\_th_n = (1 - IR\_status_n) \times BLER_{MCSx\rightarrow MCSy} + IR\_status_n \times BLER_{MCSx\_wIR\rightarrow MCSy\_wIR} \qquad (7u)$$

$$DN\_th_n = (1 - IR\_status_n) \times BLER_{MCSx \to MCSz} + IR\_status_n \times BLER_{MCSx\_wIR \to MCSz\_wIR} \quad (8u)$$

[0084] The successive step **S9** is charged to compare BLER filtered in step **S3** with the new thresholds (7u) and (8u) either coming from step **S8** or **S8'**, then in step **S10** the occurrence of an MCS switching in consequence of the previous comparisons is checked. If from the check of step **S10** it results that in step **S9** the actual BLER doesn't cross any UP_th_n or DN_th_n thresholds, the subsequent step **S11** performs an unitary increment of index n, then in step **S12** the active state of the actual TBF is monitored. Until TBF is active the respective BLER is continuously monitored from the cycle of steps **S3 - S12** to check the conditions for switching from the actual MCS; if during the cycle the TBF elapses the incoming step **S13** resets BLER and R variables and the program waits for another TBF. If during cycle **S3 - S12** the actual BLER falls below the value UP_th_n, then in step **S14** MCSx is switched to MCSy. Alternatively, if during the cycle **S3 - S12** the actual BLER exceeds the value DN_th_n, in step **S14** MCSx is switched to MCSz. When commanding the new MCS to the mobile station, in a PACKET UPLINK ACK/NACK or PACKET TIMESLOT RECONFIGURE message, the PCU unit can also set the re-segment bit to the proper value. If $IR\_status_n < 0.5$ then IR is considered as "not-properly working" and the re-segment bit is set to '1'. On the contrary, if $IR\_status_n > 0.5$ then IR is considered as "properly working" and the re-segment bit is set to '0'. For retransmissions the previous considerations are still valid so as **TABLES 5 and 6** of **APPENDIX 1.**

[0085] Whenever the Modulation and Coding Scheme is changed, in the successive step **S15** BLER and R variables are set to zero and the filtering process is re-started from step **S3.**

[0086] Additional advantage of the disclosed method is that it is independent on the memory size at the BTS. In fact if there is so much memory as the IR_status variable will always be close to 1, then in step **S9** the "perfect IR" thresholds $BLER_{MCSx\_wIR \to MCSy\_wIR}$ and $BLER_{MCSx\_wIR \to MCSz\_wIR}$ will always be used, because they are prevailing in expressions (7u) and (8u). On the contrary, if the BTS has as low memory as IR_status variable will always be close to 0, then in step **S9** the "no IR" thresholds $BLER_{MCSx \to MCSy}$, and $BLER_{MCSx \to MCSz}$ will always be used, because they are prevailing in expressions (7u) and (8u). It can be appreciated that through expressions 7u) and (8u) a sort of automatic switch between the two extreme conditions is performed.

[0087] The disclosure of how performing uplink adaptation with Incremental Redundancy carried out with reference to the **Fig.16** (the most general case), is nearly completely applicable to the downlink adaptation. Downlink adaptation is carried out by the network (BTS, BSC, PCU), as well as for uplink adaptation, but in case of downlink adaptation the receiving entities are the mobile stations which have to transmit to the network their own surveys on block decoding and the residual state of the IR memory. In practice, once the connection is established, BLER is updated at the PCU with the information provided by the EGPRS PACKET DOWNLINK ACK/NACK message, reported by the MS upon periodic request (polling) from the network. The exploitation by the PCU of the polled information suitable for calculating BLER imposes to change the time iteration index n used in the expressions (2u) and (2u') of the digital filters, and in the other descending expressions. In downlink case, time iteration index n for a given TBF connection must be replaced with reporting instant k for the same connection. So, the most general expression (2u) becomes:

$$BLER_k = f_1(BLER_{k-1}) + f_2(s_k) \quad (2d)$$

while more detailed expression (2u') requires a modification of the two weights and of the reliability variable R (expression 4u) to consider the greater lasting effect of reporting instant k. In that the following expression is valid for downlink adaptation:

$$BLER_k = \left(1 - \frac{\beta}{R_k}\right) \cdot BLER_{k-1} + \frac{\beta}{R_k} s_k \quad (2d')$$

where:

- k is the reporting instant lasting m RLC blocks;
- $$s_k = \frac{Nack\_blocks}{Sent\_blocks}$$

Nack_blocks: number of badly received RLC blocks among those sent with the present MCS.

Sent_blocks: number of blocks sent with the present MCS in the previous polling period.

- β is the forgetting factor as already defined;
- R_k denotes the reliability of the filtered BLER measurement expressed as in the following:

$$R_k = (1-\beta)^m \cdot R_{k-1} + \beta; \qquad R_{-1} = 0 \tag{4d}$$

where m is the number of radio blocks that elapsed since the last EGPRS PACKET DOWNLINK ACK/NACK message was received at the PCU. Again, R_k is the output of a running average filter that helps to keep track of the reliability of the filtered BLER measurements. In fact R_k is used to decide the weight between the new measurement (s_k) and the old measurements ($BLER_{k-1}$). When a new measurement exists (an EGPRS PACKET DOWNLINK ACK/NACK message is received), $BLER_k$ will be updated weighting new and old contributions, so to obtain the desired exponentially decreasing (with discrete time n) filter impulse response. The reliability filter is initialized at the beginning of a transmission (k=0) setting $R_{-1} = 0$. Differently from expression (4u) that uses iteration index n, expression (4d) uses iteration index k spanning several time index n, nevertheless the two expression shall perform comparable filtering function on the same filtering window, exponent m used in expression (4d) provides to this task by increasing the effect of the single iteration k in a way to opportunely dampen the old measure and reinforce the new input as if m consecutive RLC blocks were filtered in the meanwhile.

[0088] Considerations about the Incremental Redundancy, to say expression (5u) and (6u) both pertaining to IR_status and IR_check variables remain formally unchanged by using the reporting instant k. The same applies to the settlement of upgrade and downgrade thresholds through the expressions (7u) and (8u). In particular, when an EGPRS PACKET DOWNLINK ACK/NACK message is received, the MS_OUT_OF_MEMORY bit is checked:

IF

{

this bit is set (no more memory for IR is available at the MS)

}

THEN IR at instant k is considered as "not working", $IR\_check_k=0$

ELSE IR at instant k is considered as "working", $IR\_check_k=1$.

[0089] The IR status is then filtered using the same approach used for BLER:

$$IR\_status_k = f_1(IR\_status_{k-1}) + f_2(IR\_check_k) \tag{5d}$$

were function f_1 and f_2 follow the same laws as used in the BLER calculation. The analogy is extended to the most detailed function:

$$IR\_status_k = \left(1 - \frac{\beta}{R_k}\right) \cdot IR\_status_{k-1} + \frac{\beta}{R_k} \cdot IR\_check_k \tag{6d}$$

where R_k (4d) and β have already been introduced.

[0090] The IR_status variable gives information about the efficiency of Incremental Redundancy at the MS.

[0091] The linear interpolations for updating all the upgrade and downgrade tabulated BLER thresholds associated to each available MCS take now the following expressions:

$$UP\_th_k = (1 - IR\_status_k) \times BLER_{MCSx \rightarrow MCSy} + IR\_status_k \times BLER_{MCSx\_wIR \rightarrow MCSy\_wIR} \tag{7d}$$

$$DN\_th_k = (1 - IR\_status_k) \times BLER_{MCSx \rightarrow MCSz} + IR\_status_k \times BLER_{MCSx\_wIR \rightarrow MCSz\_wIR} \quad (8d)$$

where: UP_th_k and DN_th_k are the upgrade and downgrade thresholds respectively.

[0092] Additional advantages of downlink adaptation method are still those listed for uplink.

[0093] With reference to **fig.17** a modified Power Control algorithm for pursuing aims as pursued by Link Adaptation object of the present invention is now disclosed. Without limiting the invention the modified Power Control algorithm attempts to maintain a high data throughput of transmitting entities subjected to Link Adaptation with Incremental Redundancy. The modified Power Control takes part in the off-line preliminary step of link adaptation by making use of the simulation curves of net throughput (kbit/s) in function of C/I (dB) for several Modulation and Coding Schemes. The curve that grants the maximum achievable throughput (i.e. the envelope of all the curves corresponding to the different MCS in the Incremental Redundancy case) is used and reproduced in **Fig. 17** Target can be derived from the Peak Throughput QoS class requested by the mobile station. Let T_P be the Peak Throughput, then a Peak Throughput per timeslot, indicated as T_{PxTS}, is calculated:

$$T_{PxTS} = T_P / N_{TS} \quad (9)$$

where N_{TS} is the number of timeslots allocated to the TBF; i.e. N_{TS} is the minimum between the number of allocable timeslots and the number of timeslots that can be handled by the MS due to its multislot class.

[0094] Once T_{PxTS} is set on the ordinate axis of the curve "Maximum achievable throughput", the curve itself associates to the T_{PxTS} point a target C/I_{target} value on the abscissa axis. In other words the couple of points (C/I_{target}, T_{PxTS}) is marked on the "Maximum achievable throughput" curve. C/I_{target} target value constitutes the goal of the modified Power Control algorithm. Traditional Power Control algorithm attempts to minimize transmission power compatibly with a minimum fixed quality of the transmitted signal checked by the receiving entity. To reach this aim it needs to handle measures included in channel quality reports carried by associated control channels. Once the measures have been acquired, the traditional Power Control algorithm starts to increase, or decrease, step by step the transmitted power until the outlined goal on minimum quality has been checked back from the measures. Modified Power Control algorithm works as the traditional one but with a different goal, namely it tries to maintain the C/I_{target} target value for the duration of the whole TBF. The Link Adaptation algorithm subject of the present invention, on the other hand, continues to adapt to radio conditions, switching from one MCS to another, in order to optimize performance on net throughput. This may happen due to the fact that the power control cannot be "perfect" and therefore the actual C/I ratio may be different from the target one. From above it can be argued that the Modified Power Control algorithm works in synergy with the link adaptation, in that resolving the controversy outlined in the prior art.

APPENDIX 1

TABLE 1: Coding parameters for the GPRS coding schemes

Scheme	Code rate	USF	Pre-coded USF	Radio Block excl. USF and BCS	BCS	Tail	Coded bits	Punctured bits	Data rate kb/s
CS-1	1/2	3	3	181	40	4	456	0	9.05
CS-2	≈2/3	3	6	268	16	4	588	132	13.4
CS-3	≈3/4	3	6	312	16	4	676	220	15.6
CS-4	1	3	12	428	16	-	456	-	21.4

TABLE 2: Coding parameters for the EGPRS coding schemes

<table>
<tr><th>Scheme</th><th>Code rate</th><th>Header Code rate</th><th>Modulation</th><th>RLC blocks per Radio Block (20ms)</th><th>Raw Data within one Radio Block</th><th>Family</th><th>BCS</th><th>Tail payload</th><th>HCS</th><th>Data rate kb/s</th></tr>
<tr><td>MCS-9</td><td>1.0</td><td>0.36</td><td rowspan="5">8PSK</td><td>2</td><td>2x592</td><td>A</td><td rowspan="3">2x12</td><td rowspan="3">2x6</td><td rowspan="9">8</td><td>59.2</td></tr>
<tr><td>MCS-8</td><td>0.92</td><td>0.36</td><td>2</td><td>2x544</td><td>A</td><td>54.4</td></tr>
<tr><td>MCS-7</td><td>0.76</td><td>0.36</td><td>2</td><td>2x448</td><td>B</td><td>44.8</td></tr>
<tr><td>MCS-6</td><td>0.49</td><td>1/3</td><td>1</td><td>592
544+48</td><td>A</td><td rowspan="6">12</td><td rowspan="6">6</td><td>29.6
27.2</td></tr>
<tr><td>MCS-5</td><td>0.37</td><td>1/3</td><td>1</td><td>448</td><td>B</td><td>22.4</td></tr>
<tr><td>MCS-4</td><td>1.0</td><td>0.53</td><td rowspan="4">GMSK</td><td>1</td><td>352</td><td>C</td><td>17.6</td></tr>
<tr><td>MCS-3</td><td>0.80</td><td>0.53</td><td>1</td><td>296
272+24</td><td>A</td><td>14.8
13.6</td></tr>
<tr><td>MCS-2</td><td>0.66</td><td>0.53</td><td>1</td><td>224</td><td>B</td><td>11.2</td></tr>
<tr><td>MCS-1</td><td>0.53</td><td>0.53</td><td>1</td><td>176</td><td>C</td><td>8.8</td></tr>
<tr><td colspan="11">NOTE: the italic captions indicate the padding.</td></tr>
</table>

TABLE 4 – Puncturing Schemes (PS)

MCS switched from	MCS switched to	PS of last transmission before MCS switch	PS of first transmission after MCS switch
MCS-9	MCS-6	PS 1 or PS 3	PS 1
		PS 2	PS 2
MCS-6	MCS-9	PS 1	PS 3
		PS 2	PS 2
MCS-7	MCS-5	any	PS 1
MCS-5	MCS-7	any	PS 2
all other combinations		any	PS 1

APPENDIX 1

TABLE 3 - MODULATION AND CODING SCHEMES FOR EGPRS

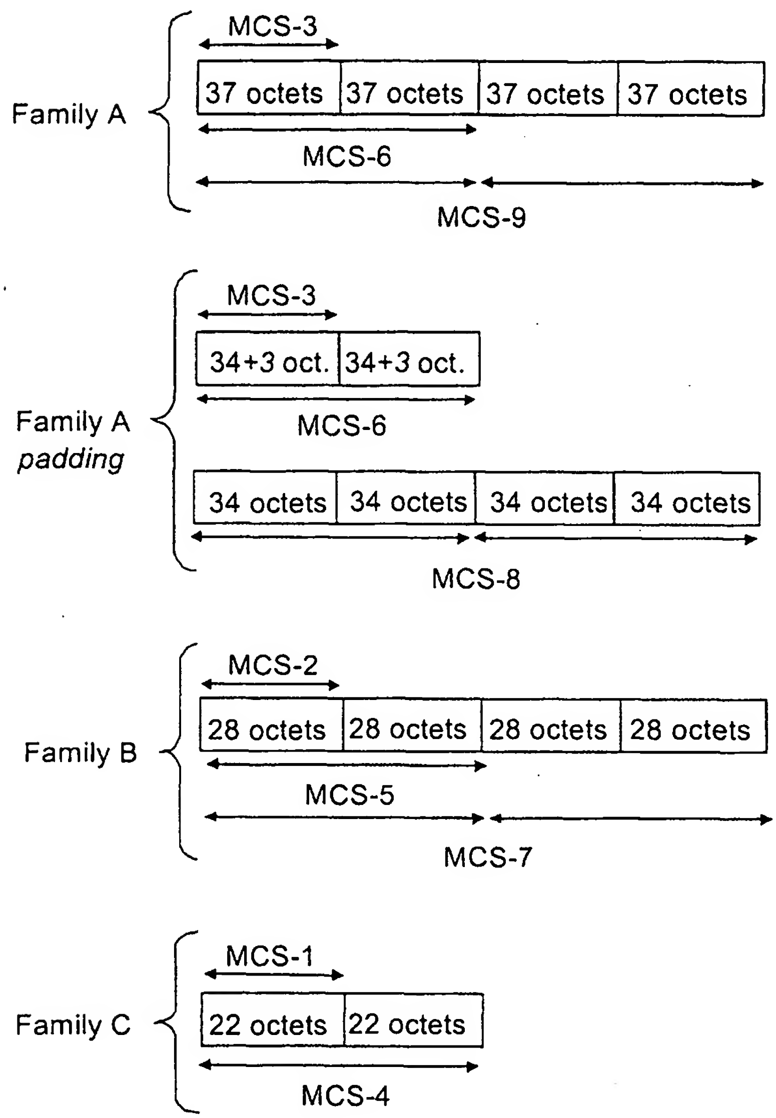


TABLE 5 - MCS to use for retransmissions when re-segmentation (re-segment bit set to '1') is carried out (specified as a function of the scheme used for the initial transmission)

Scheme used for initial transmission	Scheme to use for retransmissions after switching to a different MCS										
	MCS-9 Commanded	**MCS-8 Commanded**	**MCS-7 Commanded**	**MCS-6-9 Commanded**	**MCS-6 Commanded**	**MCS-5-7 Commanded**	**MCS-5 Commanded**	**MCS-4 Commanded**	**MCS-3 Commanded**	**MCS-2 Commanded**	**MCS-1 Commanded**
MCS-9	MCS-9	MCS-6	MCS-6	MCS-6	MCS-6	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3
MCS-8	MCS-8	MCS-8	MCS-6 (pad)	MCS-6 (pad)	MCS-6 (pad)	MCS-3 (pad)	MCS-3 (pad)	MCS-3 (pad)	MCS-3 (pad)	MCS-3 pad)	MCS-3 (pad)
MCS-7	MCS-7	MCS-7	MCS-7	MCS-5	MCS-5	MCS-5	MCS-5	MCS-2	MCS-2	MCS-2	MCS-2
MCS-6	MCS-9	MCS-6	MCS-6	MCS-9	MCS-6	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3
MCS-5	MCS-7	MCS-7	MCS-7	MCS-5	MCS-5	MCS-7	MCS-5	MCS-2	MCS-2	MCS-2	MCS-2
MCS-4	MCS-4	MCS-4	MCS-4	MCS-4	MCS-4	MCS-4	MCS-4	MCS-4	MCS-1	MCS-1	MCS-1
MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3
MCS-2	MCS-2	MCS-2	MCS-2	MCS-2	MCS-2	MCS-2	MCS-2	MCS-2	MCS-2	MCS-2	MCS-2
MCS-1	MCS-1	MCS-1	MCS-1	MCS-1	MCS-1	MCS-1	MCS-1	MCS-1	MCS-1	MCS-1	MCS-1

TABLE 6 - MCS to use for retransmissions when re-segmentation is not (re-segment bit set to '0') allowed specified as a function of the scheme used for the initial transmission)

Scheme used for initial transmis sion	Scheme to use for retransmissions after switching to a different MCS										
	MCS-9 Comm anded	**MCS-8 Comm anded**	**MCS-7 Comm anded**	**MCS-6-9 Comma nded**	**MCS-6 Comm anded**	**MCS-5-7 Comma nded**	**MCS-5 Comm anded**	**MCS-4 Comm anded**	**MCS-3 Comm anded**	**MCS-2 Comm anded**	**MCS-1 Comm anded**
MCS-9	MCS-9	MCS-6	MCS-6	MCS-6	MCS-6	MCS-6	MCS-6	MCS-6	MCS-6	MCS-6	MCS-6
MCS-8	MCS-8	MCS-8	MCS-6 (pad)	MCS-6 (pad)	MCS-6 (pad)	MCS-6 (pad)	MCS-6 (pad)	MCS-6 (pad)	MCS-6 (pad)	MCS-6 (pad)	MCS-6 (pad)
MCS-7	MCS-7	MCS-7	MCS-7	MCS-5	MCS-5	MCS-5	MCS-5	MCS-5	MCS-5	MCS-5	MCS-5
MCS-6	MCS-9	MCS-6	MCS-6	MCS-9	MCS-6	MCS-6	MCS-6	MCS-6	MCS-6	MCS-6	MCS-6
MCS-5	MCS-7	MCS-7	MCS-7	MCS-5	MCS-5	MCS-7	MCS-5	MCS-5	MCS-5	MCS-5	MCS-5
MCS-4	MCS-4	MCS-4	MCS-4	MCS-4	MCS-4	MCS-4	MCS-4	MCS-4	MCS-4	MCS-4	MCS-4
MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3
MCS-2	MCS-2	MCS-2	MCS-2	MCS-2	MCS-2	MCS-2	MCS-2	MCS-2	MCS-2	MCS-2	MCS-2
MCS-1	MCS-1	MCS-1	MCS-1	MCS-1	MCS-1	MCS-1	MCS-1	MCS-1	MCS-1	MCS-1	MCS-1

Claims

1. Method for dynamically optimize data throughput at the radio interfaces of a packet data cellular network having at disposal of said interfaces one or more type of modulations and a plurality of coding schemes for providing several combinations of modulation-and-coding schemes, or MCSs, more or less protected against transmission errors and usable for transmitting bursts of data, packed-up in blocks, between mobile stations (MS) and the serving base station (BTS), and vice versa, the throughput optimization being dynamically pursued selecting MCSs capable to grant the highest net throughput when the quality of the RF link changes starting from an initially assigned one, the selection exploiting empirical representations of the net data throughput allowable by each MCS in function of a quality parameter of the RF link, like the carrier to interference ratio, **characterized in that** it includes:

   - an off-line preliminary step for obtaining from said empirical representations some tabulated values (A, B, C, D) of Block Error Rate, or BLER, to be used like upgrade and/or downgrade thresholds associated to each available MCS for determining as many switching points between MCS having the immediately lower or higher error protection; and

   the following steps cyclically repeated for all the duration of a temporary connection set up with said initial MCS:

   - updating at each new incoming block of data an averaged value of BLER evaluated in correspondence of a commanded MCS presently in use;
   - comparison of said averaged BLER of the actual MCS with the associated upgrade and/or downgrade thresholds;
   - changing the actual MCS into said MCS immediately less error protected when the averaged BLER is lower than said upgrade threshold; or
   - changing the actual MCS into said MCS immediately more error protected when the averaged BLER is higher than said downgrade threshold.

2. Method for dynamically optimizing data throughput according to claim 1, **characterized in that** said averaged value of BLER is obtained by weighting both the preceding values of BLER and the actual decisions on errored blocks, using a digital filter having a pulse response exponentially decreasing with discrete time n spanning a block period.

3. Method for dynamically optimizing data throughput according to claim 2, **characterized in that** said pulse response of BLER digital filter is obtained by summing up two weight functions both accepting samples with the "commanded MCS", a first one to weigh the preceding values of BLER and a second one to weigh the actual decisions on errored blocks.

4. Method for dynamically optimizing data throughput according to claim 3, **characterized in that** said first and second weight functions have balanced weights, so that an arbitrary increasing of the weight of the first function also involves an equal decreasing of the weight of the second function, and vice versa.

5. M'ethod for dynamically optimizing data throughput according to claim 4, **characterized in that** the weight of said first and second weight functions are both equally varied in order to compensate the missing filtering effect of possible lacking blocks, **in that** making the outlined pulse response possible.

6. Method for dynamically optimizing data throughput according to claim 5, **characterized in that** the variation of said weights are carried out by making the said first and second weight functions further depending on a reliability function which tracks the age of the received blocks.

7. Method for dynamically optimizing data throughput according to any claim from 3 to 5, **characterized in that** said temporary connection is dedicated to transfer packet data from a selected mobile station to the base station, and said pulse response of BLER digital filter is obtained by means of the following function:

$$BLER_n = f_1(BLER_{n-1})+f_2(s_n)$$

   where:

- n is the iteration index spanning one block period;
- $s_n = 0$ if the block at instant n has been correctly received;
- $s_n = 1$ if the block at instant n has not been correctly received;
- $$s_n = \frac{1}{K}\sum_{k=1}^{K} s_{n,k}$$

  if K blocks are received for the considered connection;
- $f_1(BLER_{n-1})$ is said first weight function, taking values inside the interval 0 - 1;
- $f_2(s_n)$ is said second weight function of the variable s_n relative to the decision on the errored blocks, taking values inside the interval 0 - 1;

8. Method for dynamically optimizing data throughput according to claim 6, **characterized in that** said first and second weight functions assume the following expressions:

$$f_1(BLER_{n-1}) = (1-\beta\cdot\frac{X_n}{R_n})\cdot BLER_{n-1}$$

$$f_2(s_n) = \beta\cdot\frac{X_n}{R_n}\cdot s_n$$

where:

- x_n is equal to 1 if "at least" one RLC block for the considered connection with the commanded MCS is received at time instant n, otherwise is set to 0;
- $\beta= 1/T_{AVG}$ is a forgetting factor and T_{AVG} being the filtering period in multiples of a radio block;
- $R_n = (1-\beta)\cdot R_{n-1} + \beta\cdot x_n$; $R_{-1} = 0$ is said reliability function.

9. Method for dynamically optimizing data throughput according to any claim from 3 to 6, **characterized in that** said temporary connection is dedicated to transfer packet data from the base station to a selected mobile station, and said pulse response of BLER digital filter is obtained by means of the following function:

$$BLER_k = f_1(BLER_{k-1}) + f_2(s_k)$$

where:

- k is the reporting instant lasting m blocks;
- $$s_k = \frac{Nack\_blocks}{Sent\_blocks}$$

  Nack_blocks: number of badly received blocks among those sent with the present MCS;

  Sent_blocks: number of blocks sent with the present MCS in the previous polling period:
- $f_1(BLER_{k-1})$ is said first weight function, taking values inside the interval 0 - 1;
- $f_2(s_k)$ is said second weight function of the variable s_k relative to the decision on the errored blocks, taking values inside the interval 0 - 1.

10. Method for dynamically optimizing data throughput according to claim 8, **characterized in that** said first and second weight functions assume the following expressions:

$$f_1(BLER_{k-1}) = (1-\frac{\beta}{R_k})\cdot BLER_{k-1}$$

$$f_2(s_k) = \frac{\beta}{R_k} \cdot s_k$$

where:

- $\beta = 1/T_{AVG}$ is a forgetting factor and T_{AVG} being the filtering period in multiples of a radio block;
- $R_k = (1-\beta)^m \cdot R_{k-1} + \beta$; $R_{-1} = 0$ is said reliability function.

11. Method for dynamically optimizing data throughput according to one of the preceding claims, **characterized in that** a condition of buffer full and other main causes making retransmission with incremental redundancy inapplicable, are continuously checked on the ongoing connection at the receiver side and the relevant piece of information whether incremental redundancy is properly working or not is forwarded to the network (PCU).

12. Method for dynamically optimizing data throughput according to one of the preceding claims, **characterized in that** said tabulated values of BLER thresholds (A, B, C, D) are subdivisible into at least two former tables (A, B), a first one (A) for low-diversity RF channels and a second one (B) for high-diversity RF channels, considering as high-diversity a channel **characterized by** frequency hopping or high user mobility, and low-diversity a channel without frequency hopping and with low user mobility.

13. Method for dynamically optimizing data throughput according to claim 12, **characterized in that** each of said former tables of BLER thresholds (A, B) has a companion table (C, D) whose BLER thresholds are set taking further into account the additional effect of the incremental redundancy.

14. Method for dynamically optimizing data throughput according to claim 13, **characterized in that** all said upgrade and/or downgrade tabulated BLER thresholds (A, B, C, D) associated to each available MCS are updated at the network side (PCU) at every reception of the incremental redundancy relevant information, making use of a linear interpolation between a threshold stored into a former table (A, B) and the correspondent threshold stored into the companion table (C, D), the interpolation exploiting a network-provided variable, named for mere convenience IR_status, measuring the averaged status of incremental redundancy for the aim of unbalancing the entity of the interpolation either towards said companion table (C, D) when incremental redundancy prevails, or towards the former table (A, B) on the contrary case.

15. Method for dynamically optimizing data throughput according to claim 14, **characterized in that** said variable IR_status measuring the averaged status of incremental redundancy is obtained by weighting both the preceding values of IR_status and the actual values of a variable, named for mere convenience IR_check, taking value 1 if incremental redundancy is properly working, or value 0 on the contrary, using a digital filter having a pulse response exponentially decreasing with discrete time n spanning a block period.

16. Method for dynamically optimizing data throughput according to claim 15, **characterized in that** said temporary connection is dedicated to transfer packet data from a selected mobile station to the base station, and said pulse response of IR_status digital filter is obtained by means of the following function:

$$IR\_status_n = f_1(IR\_status_{n-1}) + f_2(IR\_check_n)$$

were:

- n is the iteration index spanning one block period;
- f_1 and f_2 are weight functions following the same laws as used in the BLER calculation.

17. Method for dynamically optimizing data throughput according to claim 16, **characterized in that** said first and second weight functions assume the following expressions:

$$f_1(IR\_status_{n-1}) = (1-\beta \cdot \frac{X_n}{R_n}) \cdot IR\_status_{n-1}$$

$$f_2(IR\_check_n) = \beta \cdot \frac{x_n}{R_n} \cdot IR\_check_n$$

where: R_n takes a formal expression as that used in the BLER calculation, while x_n and β are the same.

18. Method for dynamically optimizing data throughput according to claim 16 or 17, **characterized in that** said linear interpolation for updating all said upgrade and/or downgrade tabulated BLER thresholds associated to each available MCS take the following expressions:

$$UP\_th_n = (1 - IR\_status_n) \times BLER_{MCSx \to MCSy} + IR\_status_n \times BLER_{MCSx\_wIR \to MCSy\_wIR}$$

$$DN\_th_n = (1 - IR\_status_n) \times BLER_{MCSx \to MCSz} + IR\_status_n \times BLER_{MCSx\_wIR \to MCSz\_wIR}$$

where:

- UP_th_n and DN_th_n are said upgrade and downgrade thresholds respectively;
- $BLER_{MCSx \to MCSy}$ is an upgrade threshold stored into a said former table;
- $BLER_{MCSx\_wIR \to MCSy\_wIR}$ is an upgrade threshold stored into a said companion table;
- $BLER_{MCSx \to MCSz}$ is a downgrade threshold stored into a said former table;
- $BLER_{MCSx\_wIR \to MCSz\_wIR}$ is a downgrade threshold stored into a said companion table.

19. Method for dynamically optimizing data throughput according to claim 15, **characterized in that** said temporary connection is dedicated to transfer packet data from the base station to a selected mobile station, and said pulse response of IR_status digital filter is obtained by means of the following function:

$$IR\_status_k = f_1(IR\_status_{k-1}) + f_2(IR\_check_k)$$

were:

- k is the reporting instant lasting m blocks;
- f_1 and f_2 are weight functions following the same laws as used in the BLER calculation.

20. Method for dynamically optimizing data throughput according to claim 19, **characterized in that** said first and second weight functions assume the following expressions:

$$f_1(IR\_status_{k-1}) = (1 - \frac{\beta}{R_k}) \cdot IR\_check_{k-1}$$

$$f_2(IR\_check_k) = \frac{\beta}{R_k} \cdot IR\_check_k$$

where: R_k takes a formal expression as that used in the BLER calculation, and β is the same.

21. Method for dynamically optimizing data throughput according to claim 19 or 20, **characterized in that** said linear interpolation for updating all said upgrade and/or downgrade tabulated BLER thresholds associated to each available MCS take the following expressions:

$$UP\_th_k = (1 - IR\_status_k) \times BLER_{MCSx \to MCSy} + IR\_status_k \times BLER_{MCSx\_wIR \to MCSy\_wIR}$$

$$DN\_th_k = (1 - IR\_status_k) \times BLER_{MCSx \to MCSz} + IR\_status_k \times BLER_{MCSx\_wIR \to MCSz\_wIR}$$

where:

- UP_th_k and DN_th_k are said upgrade and downgrade thresholds respectively;
- $BLER_{MCSx \rightarrow MCSy}$ is an upgrade threshold stored into a said former table;
- $BLER_{MCSx\_wIR \rightarrow MCSy\_wIR}$ is an upgrade threshold stored into a said companion table;
- $BLER_{MCSx \rightarrow MCSz}$ is a downgrade threshold stored into a said former table;
- $BLER_{MCSx\_wIR \rightarrow MCSz\_wIR}$ is a downgrade threshold stored into a said companion table.

22. Method for dynamically optimizing data throughput according to one of the preceding claims, **characterized in that** a modified power control works in parallel with the MCS switching link adaptation and the modified power control includes the following steps:

- off-line calculation of the expression:

$$T_{PxTS} = T_P / N_{TS},$$

where: T_{PxTS} is the Peak Throughput per timeslot; T_P is the Peak Throughput derived from the Quality of Service Class of the connection, and N_{TS} is the minimum between the number of allocable timeslots and the number of timeslots that can be handled by the MS due to its multislot class;

- off-line mapping of the calculated T_{PxTS} on a simulated curve depicting the maximum achievable net throughput in function of the values of Carrier versus Interference C/I, and obtaining from the curve a target C/I_{target} value;
- exploiting the C/I_{target} for all the duration of the ongoing connection as a goal to be maintained by the network (BSC, BTS) exploiting the Power and Interference measures at the receiver side.

GSM (DCS) - GPRS (Enhanced) SYSTEM

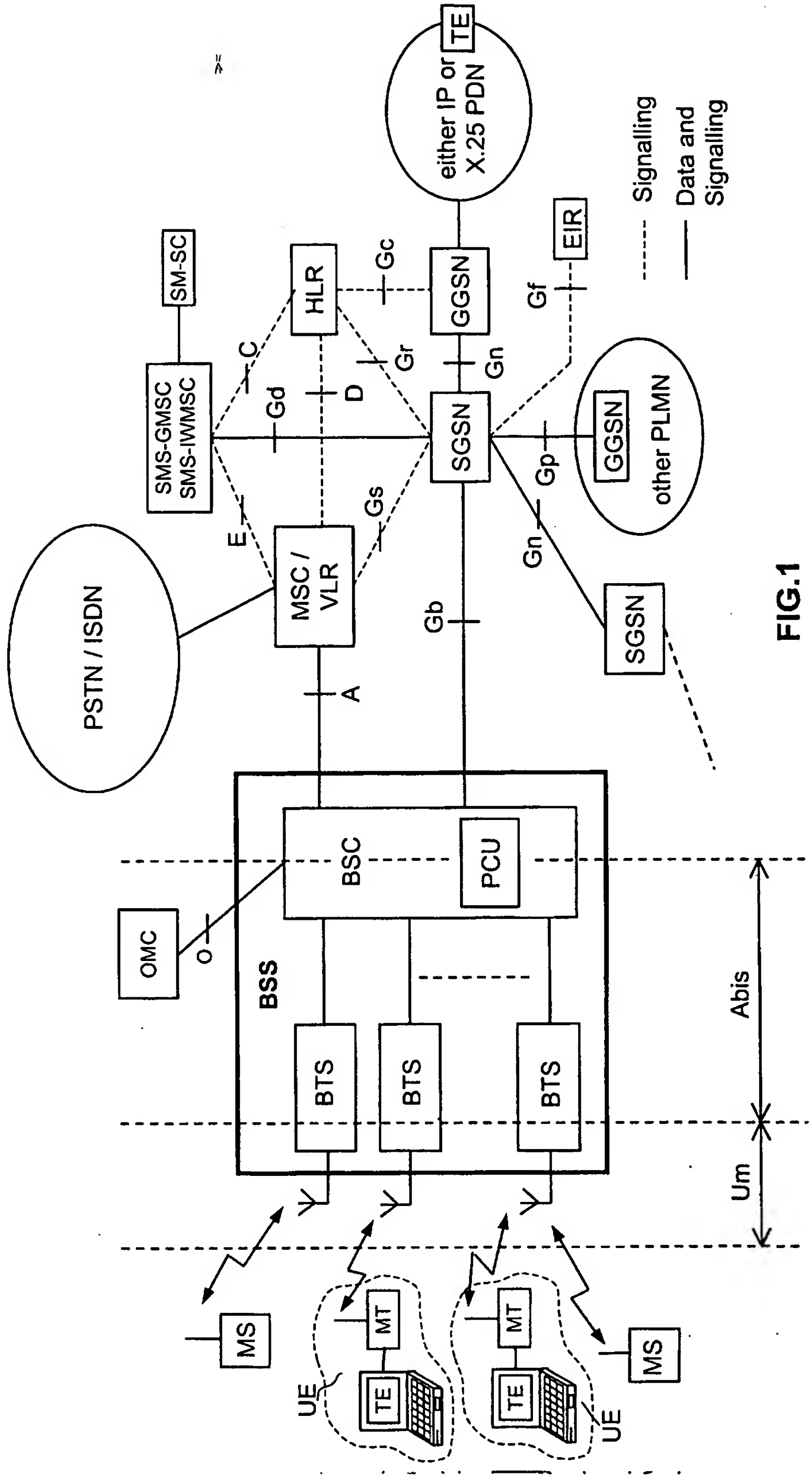


FIG.1

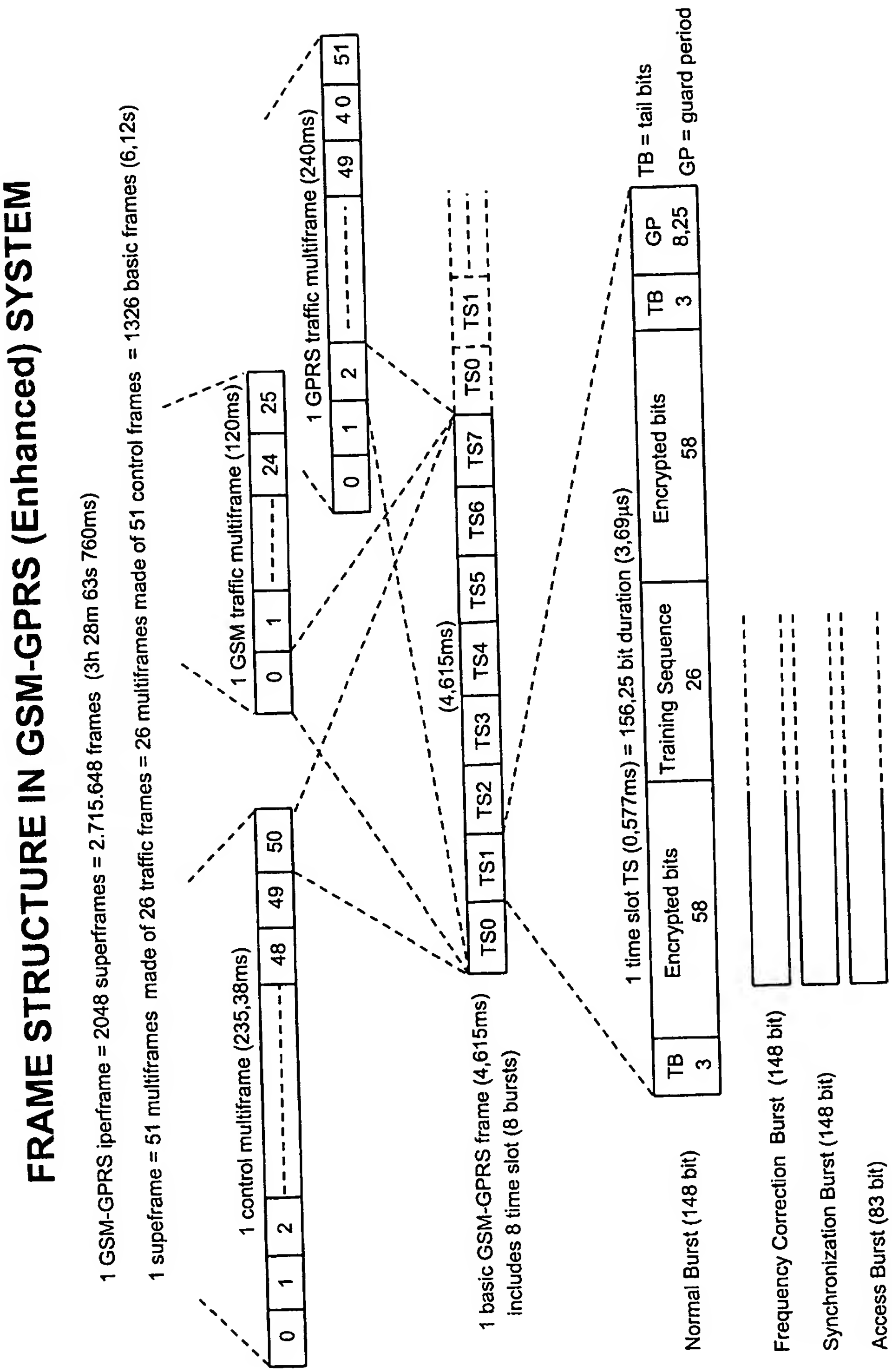

FRAME STRUCTURE IN GSM-GPRS (Enhanced) SYSTEM
1 GSM-GPRS iperframe = 2048 superframes = 2.715.648 frames (3h 28m 63s 760ms)
1 supeframe = 51 multiframes made of 26 traffic frames = 26 multiframes made of 51 control frames = 1326 basic frames (6,12s)
1 control multiframe (235,38ms)
0
1
2
48
49
50
1 GSM traffic multiframe (120ms)
0
1
24
25
1 GPRS traffic multiframe (240ms)
0
1
2
49
40
51
1 basic GSM-GPRS frame (4,615ms)
includes 8 time slot (8 bursts)
(4,615ms)
TS0
TS1
TS2
TS3
TS4
TS5
TS6
TS7
TS0
TS1
1 time slot TS (0,577ms) = 156,25 bit duration (3,69µs)
Normal Burst (148 bit)
TB
3
Encrypted bits
58
Training Sequence
26
Encrypted bits
58
TB
3
GP
8,25
TB = tail bits
GP = guard period
Frequency Correction Burst (148 bit)
Synchronization Burst (148 bit)
Access Burst (83 bit)

FIG.2

TRAFFIC CHANNEL ORGANIZATION

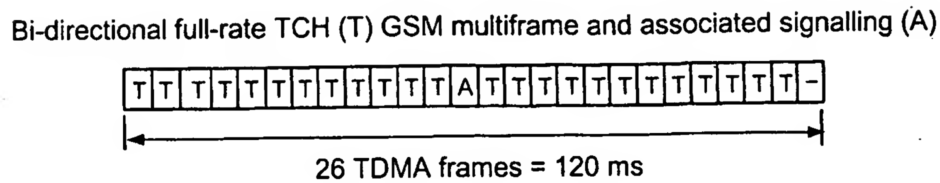


FIG.3a

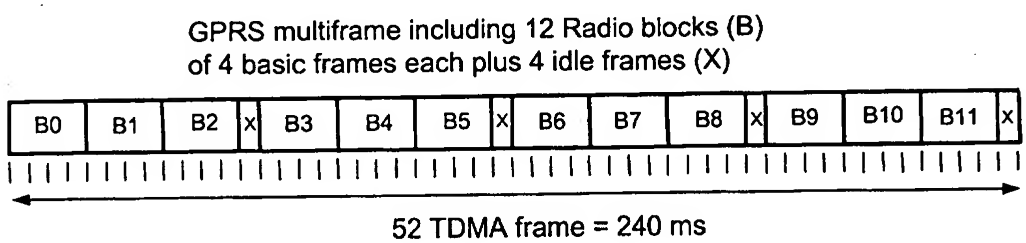


FIG.3b

MAPPING RLC LAYER INTO PHYSICAL LAYER

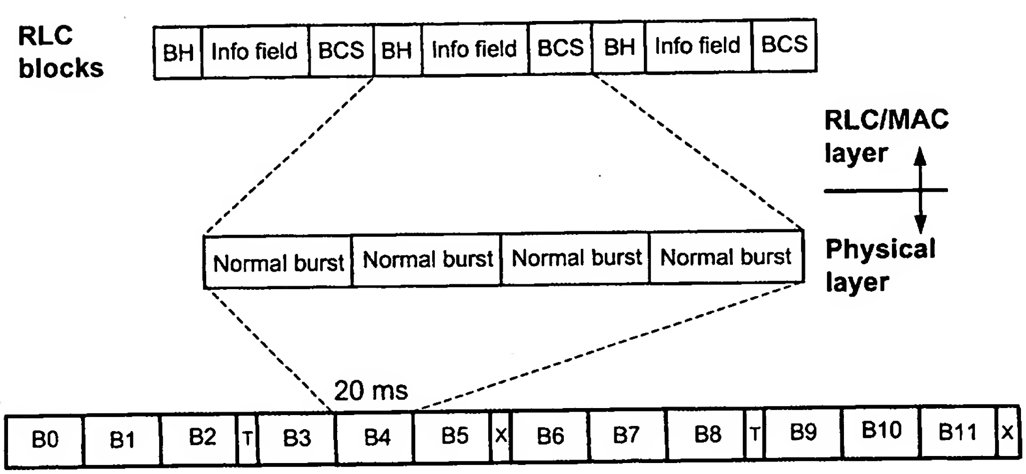


FIG.4

MOBILE STATION (MS/UE)

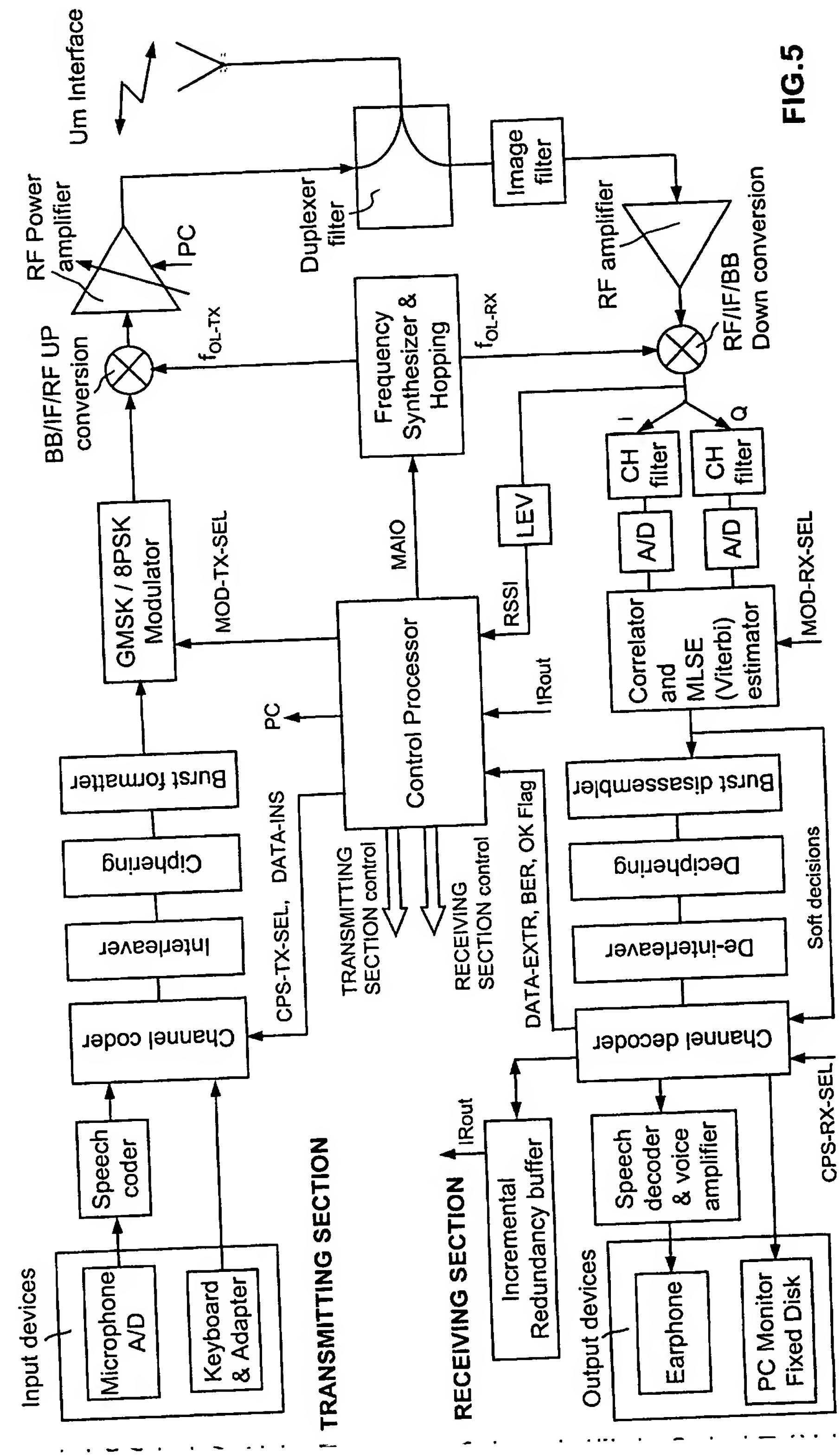
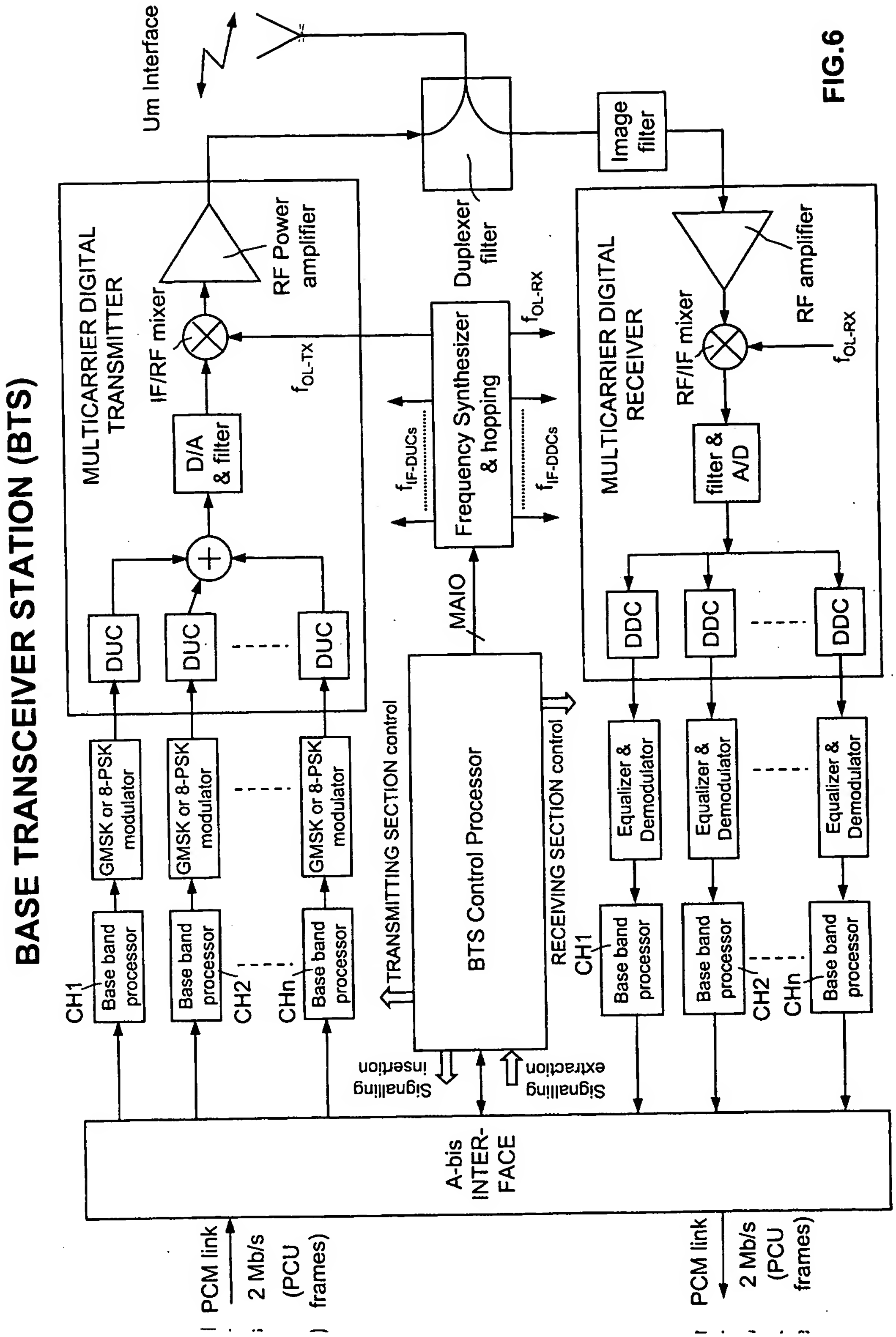


FIG.5



BASE TRANSCEIVER STATION (BTS)
Um Interface
MULTICARRIER DIGITAL TRANSMITTER
IF/RF mixer
RF Power amplifier
D/A & filter
DUC
DUC
DUC
f_{OL-TX}
Duplexer filter
Image filter
Frequency Synthesizer & hopping
$f_{IF-DUCs}$
$f_{IF-DDCs}$
f_{OL-RX}
MAIO
MULTICARRIER DIGITAL RECEIVER
RF/IF mixer
RF amplifier
f_{OL-RX}
filter & A/D
DDC
DDC
DDC
CH1
Base band processor
CH2
CHn
GMSK or 8-PSK modulator
TRANSMITTING SECTION control
BTS Control Processor
RECEIVING SECTION control
Equalizer & Demodulator
Signalling insertion
Signalling extraction
A-bis INTER-FACE
PCM link
2 Mb/s (PCU frames)

FIG.6

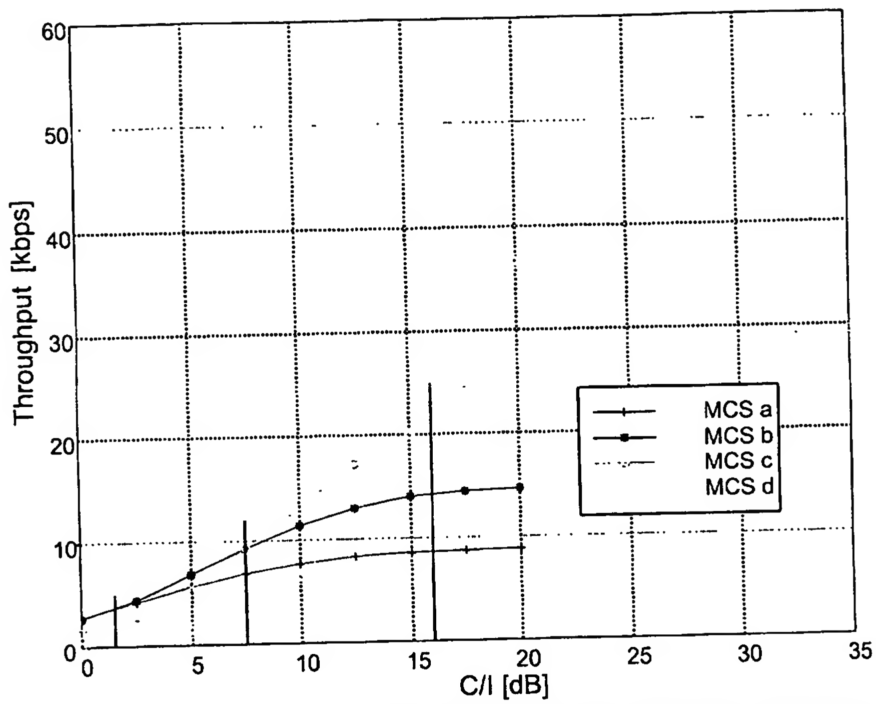


FIG.7 Simulation results for a selection of MCS (low diversity, without IR)

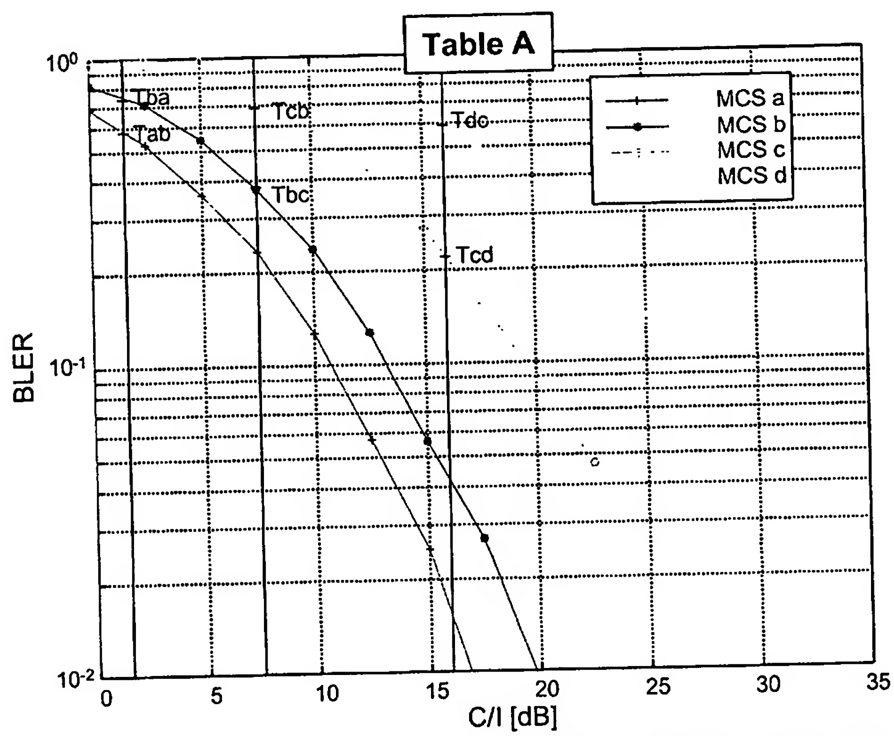


FIG.8 BLER versus C/I for a selection of MCS (low diversity, without IR)

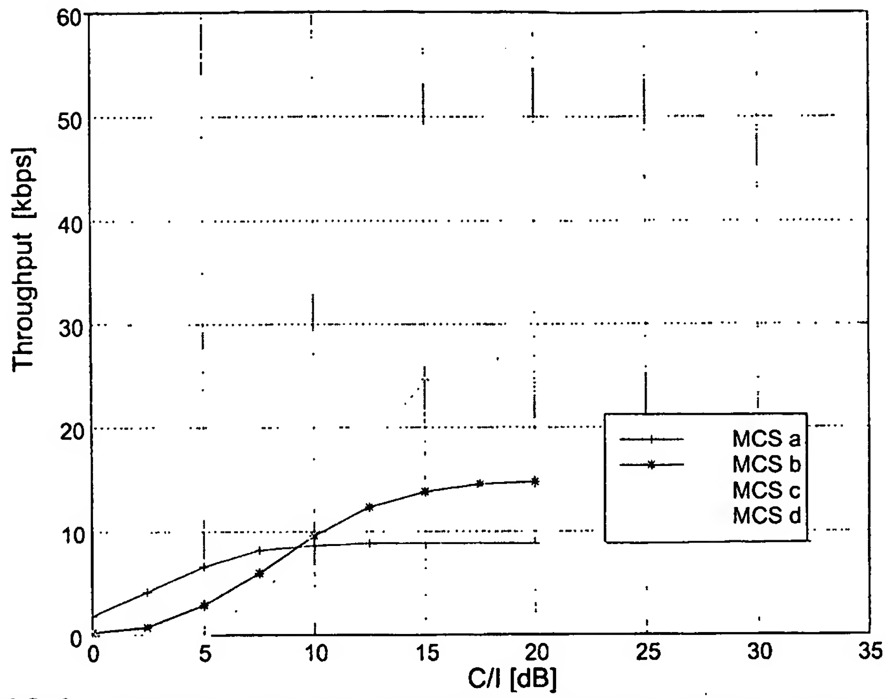


FIG.9 Simulation results for a selection of MCS (high diversity, without IR)

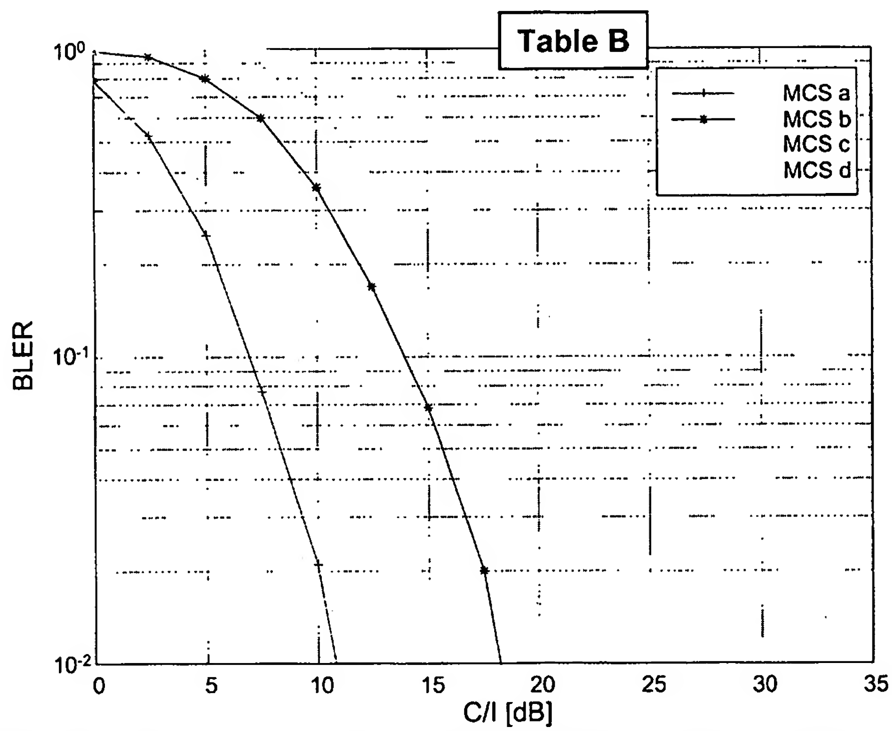


FIG.10 BLER versus C/I for a selection of MCS (high diversity, without IR)

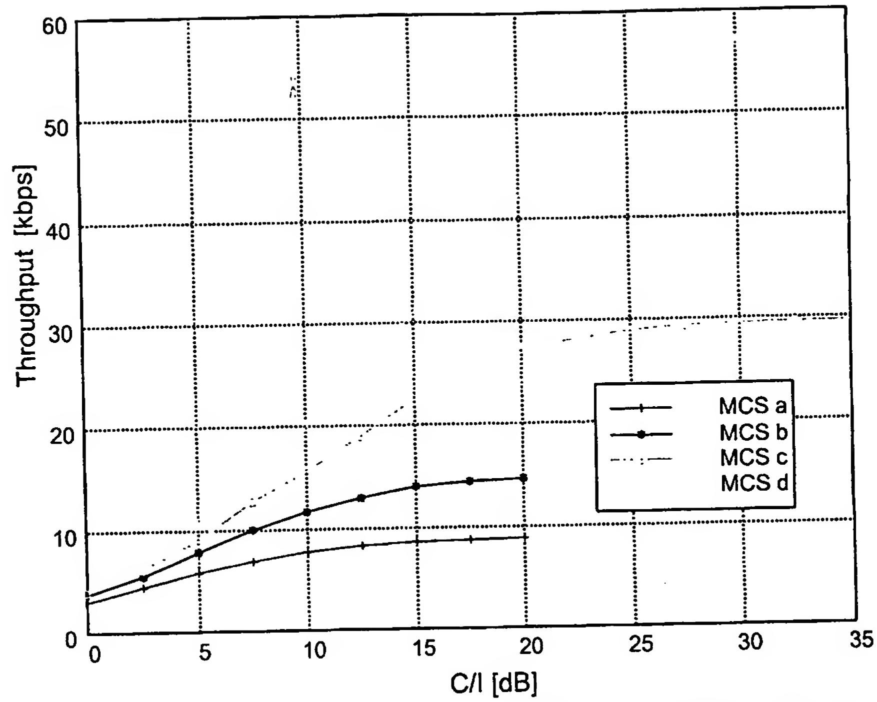


FIG.11 Simulation results for a selection of MCS (low diversity, with IR)

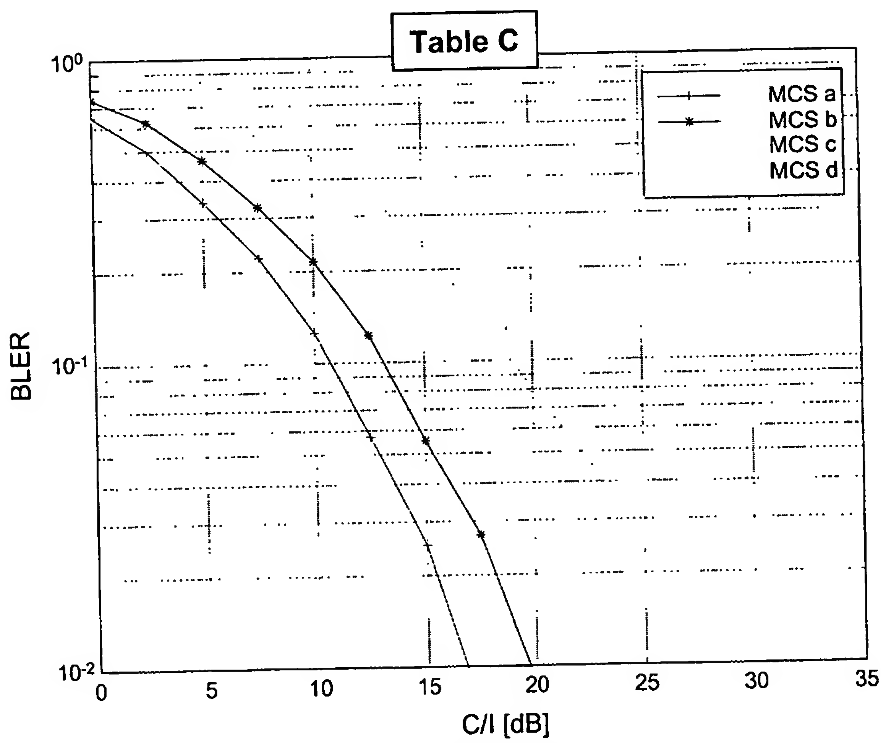


FIG.12 BLER versus C/I for a selection of MCS (low diversity, with IR)

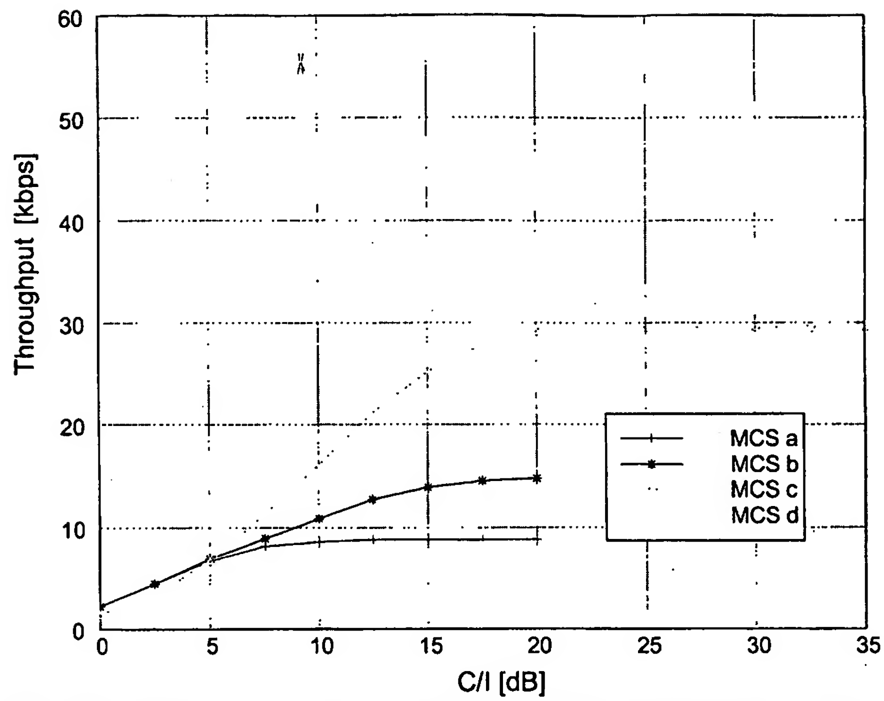


FIG.13 Simulation results for a selection of MCS (high diversity, with IR)

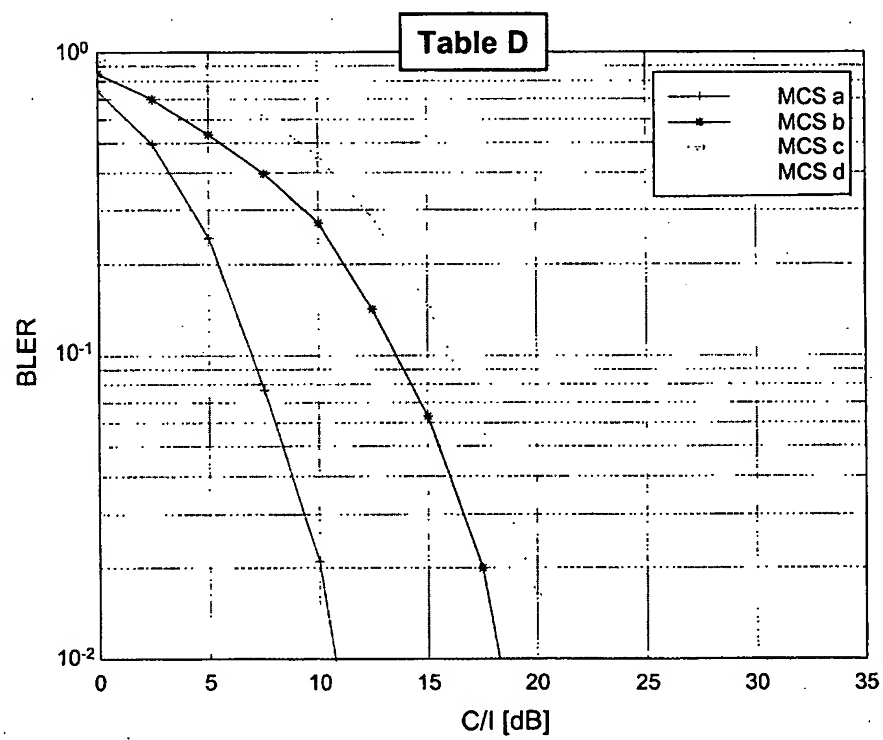


FIG.14 BLER versus C/I for a selection of MCS (high diversity, with IR)

METHOD TO PERFORM LINK ADAPTATION WITHOUT IR

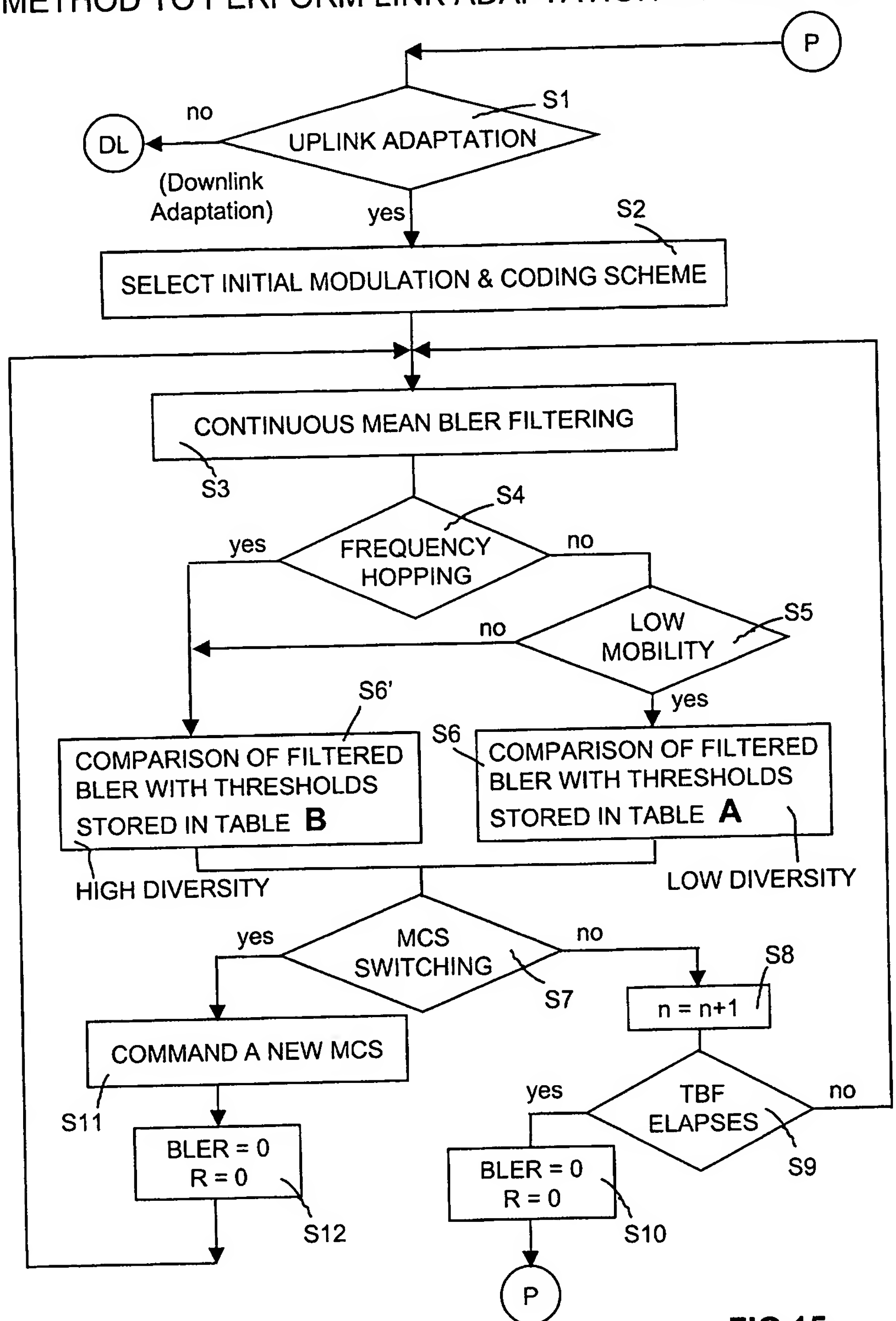


FIG.15

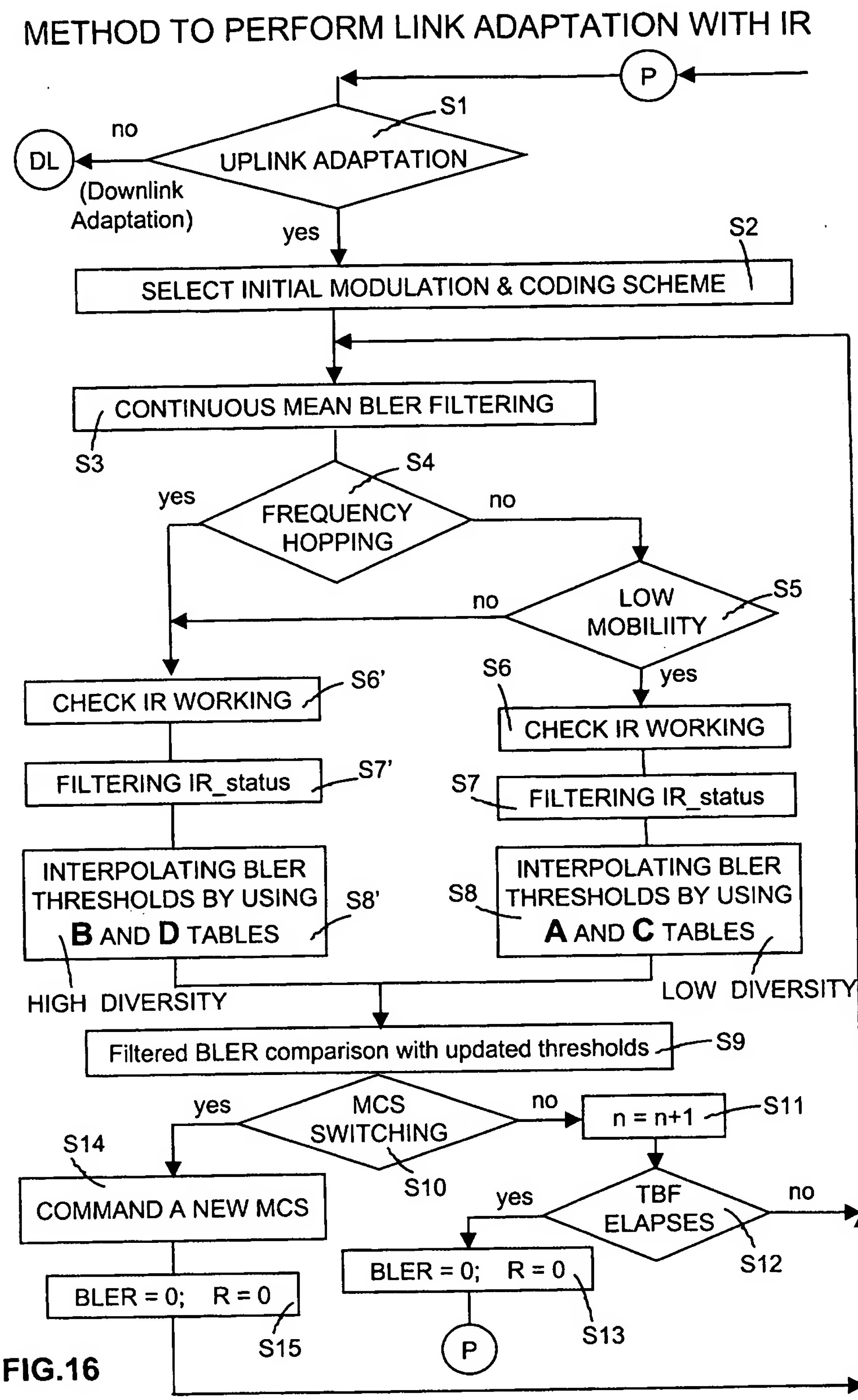
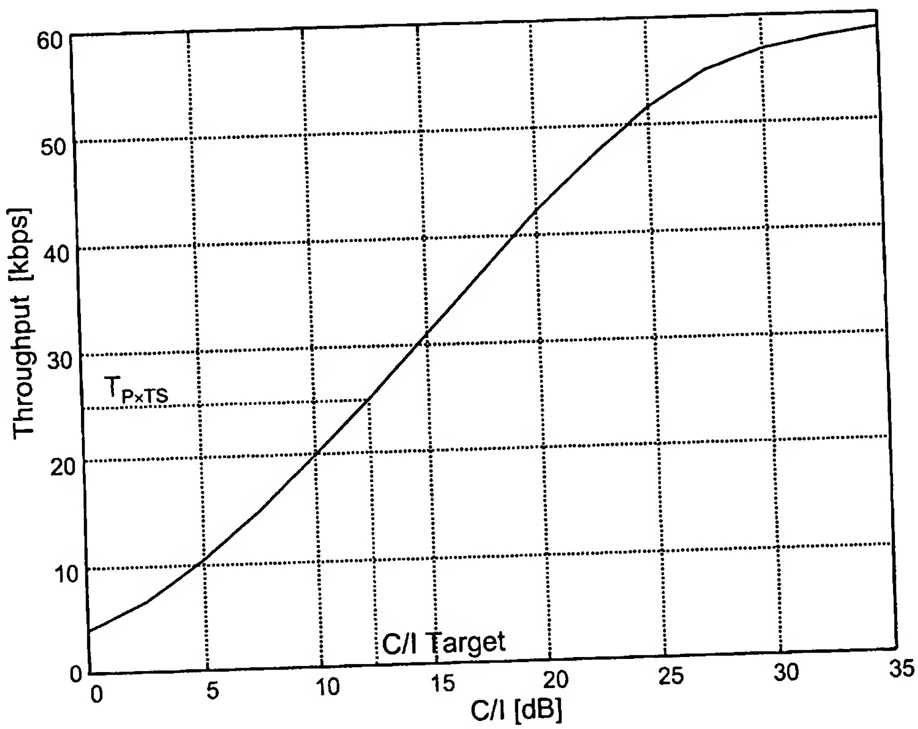

METHOD TO PERFORM LINK ADAPTATION WITH IR
P
S1
no
DL
UPLINK ADAPTATION
(Downlink
Adaptation)
yes
S2
SELECT INITIAL MODULATION & CODING SCHEME
CONTINUOUS MEAN BLER FILTERING
S3
S4
yes
FREQUENCY
HOPPING
no
LOW
MOBILIITY
S5
no
yes
S6'
CHECK IR WORKING
S6
CHECK IR WORKING
FILTERING IR_status
S7'
S7
FILTERING IR_status
INTERPOLATING BLER
THRESHOLDS BY USING
B AND D TABLES
S8'
S8
INTERPOLATING BLER
THRESHOLDS BY USING
A AND C TABLES
HIGH DIVERSITY
LOW DIVERSITY
Filtered BLER comparison with updated thresholds
S9
yes
MCS
SWITCHING
no
n = n+1
S11
S14
COMMAND A NEW MCS
S10
yes
TBF
ELAPSES
no
BLER = 0; R = 0
S12
BLER = 0; R = 0
S15
P
S13

FIG.16



Maximum achievable throughput (with IR)

FIG.17

European Patent Office

EUROPEAN SEARCH REPORT

Application Number
EP 01 83 0283

DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (Int.Cl.7)
X Y	US 5 764 699 A (CRISLER KENNETH J ET AL) 9 June 1998 (1998-06-09) * abstract * * figures 3,4 * * column 5, line 59 - column 7, line 27 * ---	1 2-4,7,9	H04L1/00
Y	SHELDON M. ROSS: "Introduction to Probability and Statistics for Engineers and Scientists" 1987 , JOHN WILEY&SONS , NEW YORK XP002180737 * page 426 - page 427 * ---	2-4,7,9	
A,D	WO 99 12304 A (ERICSSON TELEFON AB L M) 11 March 1999 (1999-03-11) * the whole document * -----	1-10	

TECHNICAL FIELDS SEARCHED (Int.Cl.7)
H04L

~~The present search report has been drawn up for all claims~~

Place of search	Date of completion of the search	Examiner
THE HAGUE	29 January 2002	Borges, P

CATEGORY OF CITED DOCUMENTS

X : particularly relevant if taken alone
Y : particularly relevant if combined with another document of the same category
A : technological background
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P : intermediate document

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E : earlier patent document, but published on, or after the filing date
D : document cited in the application
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ANNEX TO THE EUROPEAN SEARCH REPORT ON EUROPEAN PATENT APPLICATION NO.

EP 01 83 0283

This annex lists the patent family members relating to the patent documents cited in the above-mentioned European search report.
The members are as contained in the European Patent Office EDP file on
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29-01-2002

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
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WO 9912304 A	11-03-1999	US 6167031 A	26-12-2000
		AU 9012598 A	22-03-1999
		BR 9811397 A	22-08-2000
		CA 2300884 A	11-03-1999
		CN 1277764 T	20-12-2000
		EP 1010288 A	21-06-2000
		JP 2001515307 T	18-09-2001
		TW 410514 B	01-11-2000

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For more details about this annex : see Official Journal of the European Patent Office, No. 12/82

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(11) EP 1 255 368 A1

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Palazzo Gorky
Via Monfalcone, 1
20092 Cinisello Balsamo (IT)

(54) Method to perform link adaptation in enhanced cellular communication systems with several modulation and coding schemes

(57) Method to perform link adaptation at the radio interfaces of an enhanced packet data cellular network handling several Modulation and Coding Schemes (MCS) for maximizing data throughput. In a preliminary off-line step behavior in terms of net throughput of the various available MCSs is simulated for different C/I conditions. From the simulation two sets of tables are obtained, each table including upgrade and downgrade thresholds expressed in terms of Block Error Rate (BLER). Thresholds correspond to switching points from an MCS to the two available MCSs having the immediate less or more protection. The two sets of tables are referred to higher or lower diversity RF environments and are further specialized for taking into account EGPRS type II hybrid ARQ, namely Incremental Redundancy (IR). During transmission the transmitted blocks are checked for FEC and the results are sent to the network. The network continuously updates BLER using exponential smoothing. In order to achieve the correct time response, in spite of that RLC blocks can be received or not, a reliability filter is provided whose output is used to decide the weight between the new and old measurements to make the BLER filter impulse response exponentially decreasing with time. The IR efficiency is tested for each incoming block and an indicative variable IR_status is filtered using the same approach used for BLER. Each actual threshold of BLER to be used in link adaptation is obtained by a linear interpolation between the tabulated threshold without IR and with perfect IR, both weighed with filtered IR_status. Filtered BLER is then compared with said interpolated thresholds for testing the incoming of a MCS switching condition. Power control pursues the goal of maintaining constant QoS peak throughput per time slot.

METHOD TO PERFORM LINK ADAPTATION WITH IR

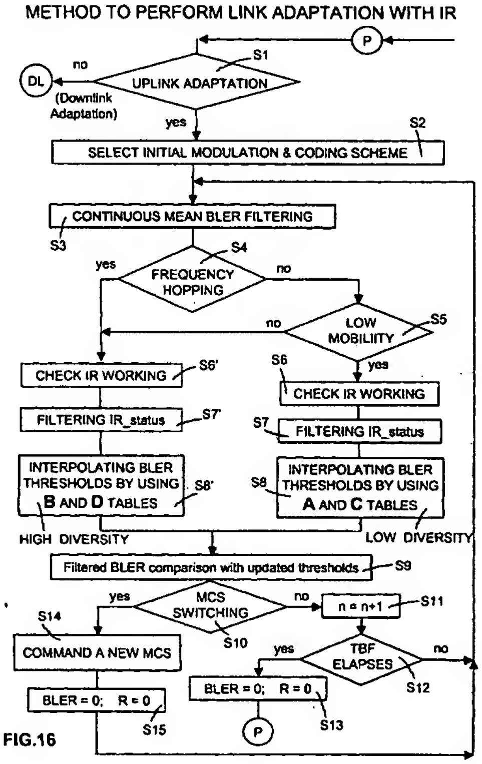

FIG.16

EP 1 255 368 A1

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Description

FIELD OF THE INVENTION

[0001] The present invention relates to the field of radiomobile communication systems and more precisely to a method to perform link adaptation in enhanced cellular communication systems with several modulation and coding schemes.

BACKGROUND ART

[0002] A significant example of known art in the above technical field is disclosed in patent application WO 99/12304 filed by ERICSSON, titled: A METHOD FOR SELECTING A COMBINATION OF MODULATION AND CHANNEL CODING SCHEMES IN A DIGITAL COMMUNICATION SYSTEM". Both the invention disclosed in the cited document and the invention in subject, that will be disclosed later on, are suitable to be employed in the so-called General Packet Radio Service (GPRS) recently added to the Global System for Mobile communications (GSM) for enabling it to manage packet data. So an introduction of the GSM-GPRS system is needed before discussing the apparently nearest prior art. The introduction takes advantage from the large **GSM** standardization coming from ETSI (European Telecommunications Standards Institute) and also from the volume titled: "The GSM System for Mobile Communication", edited by the authors: Michel MOULY, Marie-Bernadette PAUTET, Copyright 1992).

[0003] **Fig.1** of the description is similar to the fig. 2 of standard ETSI **GSM 03.60** - Service description. The system of fig.1 represents a cellular GSM(DCS)-GPRS(Enhanced) network including mobile stations communicating via radio with a fixed remaining part. In fig.1 are visible first type of Mobile Stations MS suitable for voice communication (and short messages) and second type of mobile stations named User Equipment UE each comprised of a Terminal Equipment TE for handling data (as a PC) connected to a Mobile Terminating equipment MT suitable to data packet transmission. Mobile stations MS and UE camped on a cell are connected via standard on air interface Um to a fixed Base Transceiver Station BTS which serves either a central or trisectorial cell belongs to a clustered geographical area covered by GSM-GPRS Public Land Mobile Network PLMN. In fig.1 more base stations BTS are connected to a Base Station Controller BSC through a not fully standardized Abis interface. The BSC controller includes a block PCU (Packet Control Unit) relevant for the present invention. The BSC controller and the interconnected base station BTS constitute a Base Station Subsystem BSS serving a cluster of cells. An BSC controller in its turn is connected to a Message Switching Centre MSC and to a Service GPRS Support Node SGSN via standard interfaces A and Gb respectively, both supporting SS7 signalling. The MSC centre and SGSN node are connected to a Home Location Register HLR and a Visitor Location Register VLR which add intelligence to the network by allowing mobility of communications. The MSC centre and SGSN node support Short Message Service SMS, being for this purpose connected to a Short Message Service Centre SM-SC via the functions SMS-GMSC (Short Message Service - Gateway MSC) and SMS-IWMSC (SMS - InterWorking MSC). The SGSN node is further connected to: 1) another SGSN node of the same PLMN network through a standard Gn interface; 2) a Gateway GSN node GGSN belonging to another PLMN network through a standard Gp interface; 3) a Gateway GSN node GGSN belonging to the same PLMN network, through the Gn interface, and the GGSN node is connected to either an IP (Internet Protocol) or X.25 Public Data Network PDN specialized in packet data routing; 4) finally to an Equipment Identity Register EIR. The MSC centre is connected to the Public Switching. Telephone Network PSTN also comprised of an Integrated Services Digital Network ISDN. Besides the mentioned interfaces also the following standard ones are provided: Gf, Gs, Gr, Gd, D, E, C whose connections are visible in fig.1.

[0004] The schematized GSM-GPRS system is capable to switch both the traditional voice and data circuits and the new packet data which don't request fixed connections for all the duration of an active session. The SGSN node has the same role for packet data as the MSC centre has for voice circuits, it traces individual locations of the mobile stations enabled for data packet communication and performs security and access control functions. For this purpose the HLR register includes information concerning GPRS users. The GGSN node provides interworking with external data packet switching networks, in particular with a backbone network based on IP protocol.

[0005] Both GSM and GPRS use standard procedures at the relevant interfaces, namely for: synchronization, cell selection and reselection, paging, access control, request a dedicated channel, security, error detection and correction, retransmission, power control, voice and data flux control, routing, handover, billing, etc. Such procedures belong to a most general protocol having a layered structure named "Transmission Plane" proposed by the International Organization for Standardization (ISO) for Open System Interconnection (OSI). Based on ISO model an OSI system can be described by means of a set of subsystems fit in a protocol stack. A subsystem N which consists of one or more entities of level N interacts only with subsystems immediately upon and below it and a level N entity operates into its own level N. Peer level N entities communicate each other by using services from the underlying layer N. Similarly, layer N services are provided to the layer N+1 at an N-Service Access Point named N-SAP. Information transferred from a starting to an arrival point is always conveyed by physical channels provided at the crossed interfaces. Relevant layers

for the arguments developed in this disclosure are the following:

- Radio Link Control / Medium Access Control (RLC/MAC). The RLC layer-2 function provides a radio link with reliability and maps into GSM physical channels the Link Layer Control (LLC) layer-3 frames. The MAC function is provided to control and signalling procedures for accessing radio channel, i.e. request and grant. RLC/MAC protocol is standardized in **GSM 04.60.**
- GSM RF is pertaining to the physical radio channel at the Um interface as standardized in the series of specifications **GSM 05.xx** The physical channel relevant for GPRS service is named PDCH (Packet Data Channel).

[0006] At GPRS planning stage the compatibility with pre-existent GSM has been deliberately maintained to enable GPRS of exploiting the same physical channels as GSM at the Um interface and consequently promoting an easy integration. Both for GSM and GPRS there are signalling channels and traffic channels, the first ones are either for broadcast common control or for dedicated control, the second ones are either for voice or packet data. The additional logical GPRS channels, although referred to packet data have names and functional characteristics which follow from the conventional GSM channels; examples of relevant GPRS channels are the following: PBCCH (Packet Broadcast Control Channel), PCCCH (Packet Common Control Channel), PACCH (Packet Associated Control Channel), e PDTCH (Packet Data Traffic Channel). A list of relevant channels is reported in the specification **GSM 05.01** titled "Physical layer on the radio path".

[0007] The Extended GSM 900 system is required to operate in the following frequency bands:

- 880 - 915 MHz: mobile stations transmit uplink, base station receives;
- 925 - 960 MHz: base station transmits downlink, mobile stations receive; while for Digital Cellular System DCS 1 800 the system is required to operate in the following frequency bands:
- 1.710 - 1.785 MHz: mobile stations transmit uplink, base station receives;
- 1.805 - 1.880 MHz: base station transmits downlink, mobile stations receive.

Each of the above frequency band is also used in GPRS service and includes a plurality of modulated carriers spaced 200 kHz apart. Full-duplex communications take place by Frequency Division Duplexing (FDD) technique. A carrier among those in use in a cell is assigned for all the duration of a timeslot TS out of eight cyclically repeated to allow time division among the users. During the assigned timeslot Modulation (detailed in **GSM 05.04)** impresses the characteristics of the modulating burst onto one or more physical parameters of a digital carrier to be transmitted at radiofrequency. The GSM-GPRS system exploits a GMSK (Gaussian Minimum Shift Keying) modulation that is a non-linear Continuous Phase Modulation (CPM) characterized by compact spectrum and constant modulation envelope. Compact spectrum generates poor interferences into adjacent frequency channels by introducing a slight worsening of the intersymbolic interference. Constant modulation envelope allows the gain saturation of the power amplifier (class C amplifying) and consequent energy saving from the power supply. Besides power control becomes simpler.

[0008] With reference to **fig.2** it can be appreciate the sequential organization of 8 timeslots TS0, ..., TS7 constituting a 4,615 ms basic frame used in Time Division Multiple Access (TDMA) GSM-GPRS system. Four different typologies of burst are provided corresponding to the possible contents of any timeslot. The sequential frames are organized within more hierarchical levels observed by all the carriers used in the system. All the carriers transmitted by a BTS have reciprocally synchronized frames. Starting in the figure from bottom to top each timeslot has 0,577 ms duration, corresponding to 156,25 $\times$ 3.69 μs bit duration, and carries an information burst containing 142 useful bits, 3+3 tail bits TB, and a guard time GP without information 8,25 bits long. The 3.69 μs bit duration corresponds to 270,83 kbit/s which is the system cipher rate. The burst can be of four different types, namely: Normal burst, Frequency Correction burst, Synchronization burst, and Access burst. For the purposes of disclosure the only Normal burst is depicted in fig. 2 where it includes 2$\times$58 useful bits, redundancy included, and 26 bits of a training sequence in midamble position. Training sequence is a known pattern used to dynamically synchronize the received burst and to estimate the impulse response of the radio channel for correctly demodulating the incoming signal. The nature of the 116 bits payload will be detailed later on, distinguishing between GSM and GPRS. Continuing towards the upper part of **fig.2** it can be noticed that two different typologies of multiframes are foreseen, namely a signalling multiframe for carrying control channels and a traffic multiframe for carrying payloads and associated signalling. The signalling multiframe is 253,38 ms long and includes 51 basic TDMA frames. A GSM traffic multiframe is 120 ms long and includes 26 basic TDMA frames. A GPRS traffic multiframe is 240 ms long and includes 52 basic TDMA frames. The two type of multiframes concur to form a unique superframe 6,12 seconds long, consisting of 1326 basic TDMA frames, finally 2048 sequential superframes form one iperframe of 2.715.648 basic frames TDMA of 3h 28m 63s 760ms duration. A frame Number FN referred to the frame position in the iperframe is broadcasted within the cell.

[0009] **Figures 3a and 3b** show traffic channel organization in the TDMA multiframes for voice/data and packet data respectively. **Fig.3a** concerns GSM payload where a multiframe of 26 basic frame includes: 24 traffic frames (T), 1

associated control frame (A), and 1 idle frame (-). A physical channel inside a multiframe is constituted by the combination of one frequency and one repetitive time slot. A burst of fig.

[0010] 2 generates a period of RF carrier which is modulated by the relevant data stream. A burst therefore represents the physical content of a timeslot.

[0011] **Fig.3b** concerns GPRS payload where a multiframe of 52 basic frame includes 12 radio blocks B0, ... B11 of 4 basic frames each, intercalated with an idle frame (X) every three radio blocks. A radio block is carried on a channel defined as above spanning over 4 TDMA frames, so as the mean transmission time of a RLC block is near 20 ms.

[0012] **Fig.4** is referred to the GPRS service and shows a mapping of sequential RLC layer blocks into physical layer. Each RLC block includes a block header BH of variable length, an information field comprising data coming from the upper layer LLC, and a field Block Check Sequence BCS used for error detection. A single RLC block is mapped into 4 sequential frames of the TDMA multiframe. So until 8 users can be interleaved in the period of a radio block.

[0013] GSM's payload timeslots are allocated one to one to the different users, both in uplink and downlink, while as far as concerns GPRS service a flexible allocation is available. More precisely: 1) GPRS's payload timeslots are independently allocated in uplink and/or downlink; 2) singular users can take advantage of multislot allocation; 3) each configured data packet physical channel PDCH (timeslot) can be advantageously shared among different users which access it on the basis of appropriate priority rules. The MAC layer of GPRS protocol has appropriate procedures for governing dynamic allocation of the resources. Control messages to set up or set down a connection activate said procedures for packet data transfer. Temporary Block Flows (TBF) are connections set up on physical layer by the MAC procedures, they include memory buffers to accommodate the queues of RLC/MAC radio blocks. Each TBF connection allows unidirectional point-to-point transfer of user data and signalling between a mobile station and base station, or vice versa. A TBF connection is held for the only transfer of all the RLC/MAC blocks of a LLC protocol session. The network assigns to each TBF connection a respective Temporary Flow Identity named TFI identifier by associating a field in the header of RLC/MAC blocks. The mobile stations shall assume that TFI identifier is unique for uplink or downlink concurrent TBFs (i.e. assigned to the same MS/UE). The header of RLC/MAC blocks further includes fields to specify direction and type of a control message.

[0014] In case of dynamic allocation of the resources and in presence of at least one uplink TBF connection, the header of each RLC/MAC block transmitted downlink includes an Uplink State Flag field (3 bits) named USF written from the network to enable the uplink transmission of a successive radio block from one out M mobile stations which share the same uplink PDCH channel.

[0015] GSM-GPRS system bears three classes of operation for mobile stations: a class A mobile operates with GSM and GPRS simultaneously; a class B mobile watches GSM and GPRS control channels but can operate only a set of service at a time; finally a class C mobile only uses GPRS services. Furthermore physical resources at the Um interface can be shared between speech and packet data services on the basis of traffic charge at the initial cell planning.

[0016] GPRS service bears Quality of Service (QoS) to assure among other things the following requirements: respect of a negotiated priority of service, service reliability, guarantee of a fixed end-to-end packet transfer delay, guarantee of medium and peak throughput in conformity with a certain multi-slot class. QoS parameters together with A, B, and C classes of operation and class of multislot capability take part in a User Profile made known to the network during GPRS attach.

[0017] A generic cellular telephony system suffers a lot of impairments mainly due to the following causes:

1. The peculiarity of radio propagation in conformity with the typology of the cells.
2. The mobility of the users
3. The intrinsic frequency reuse.

[0018] An impairment due to the first cause of above is the time dispersive behavior of the propagation medium because of non-linearities which distort the original shape of transmitted pulses, causing intersymbol interference due to the pulse spreading over adjacent symbol intervals. Another impairment descending from the same cause is undoubtedly multipath fading due to the random presence of spotted atmospheric diffusers on the radio path introducing statistical behavior on the radio propagation. Both at MS or BTS receiving antennas, various phase-shifted echoes of a transmitted signal coming from multipath are summed up with random distributed phases. The result is an amplitude envelope attenuated below certain levels during corresponding fade durations taken as observation times. The time-varying fading behavior is a statistical process whose probability density follows the Rayleigh distribution. Multipath fading is spectrally characterized to be either flat or frequency selective (notch), this happen respectively for correlated or uncorrelated scattering and in both the cases it generates burst errors. Last shortcoming of an on air interface is its vulnerability due to the easiness of malicious interception of data and conversations, if not otherwise provided.

[0019] Impairments due to the user mobility mainly are: shadow fading (i.e. corner), Doppler effect, and a certain spreading of Time Of Arrivals (TOA) of RF signals at the BTS antenna because of the various distances of the mobile

stations. Shadow fading is caused by the incoming of an obstacle along the line of sight propagation. By shadow fading the transmitted RF signal undergoes an additional steep attenuation to the usual path attenuation. Doppler effect is a slight frequency shift proportional to the speed the mobile station; the shift introduces noise phase and makes time-variant the channel response. Doppler effect for the highest speeds disturbs the synchronization process and the estimate of the channel pulse response consequently. The spreading of the TOAs force the realignment of the RF signals at the BTS side.

[0020] An impairment due to the frequency reuse is the presence of isofrequential interferent signals coming from the neighbor cells. C/I ratio increases and the quality of the reconstructed signal gets worsen consequently. The smaller are the cells the greater is the allowed traffic throughput but parallely cochannel interference increases due to the heavy frequency reuse.

[0021] The finding of strategies effective to neutralize the above causes of misoperation in cellular systems, needs a good knowledge of the various electromagnetic environments. Typical Radiomobile channels have been extensively studied and experimented with large, medium, and small cells. Small urban cells are further subdivided into micro and pico cells (i.e. canyons). Large and medium cells are subjected to a variety of environments, such as: hilly terrain, mountainous, woody, motorway, urban, etc. Starting from the above considerations ETSI standard committee has specified in **GSM 05.05** some practical pulse responses for typical radiomobile channels, such as: Hata-Okumura, COST231, TU3, TH, etc. Most recently models have been proposed in which the RF channel is also spatially characterized.

[0022] A lot of countermeasures have been introduced into the cellular systems to combat the above drawbacks; the most popular are the following: channel coding - interleaving - ciphering - channel equalization - frame alignment - frame (block) retransmission - slow frequency hopping - power control - intra-cell handover - and lastly link adaptation. They are valid in general so that speech, traffic data, and signalling can take advantage from them. Obviously link adaptation is the countermeasure that mostly impacts the present invention: it can be specialized into speech or data adaptation. Recently link adaptation has been improved in concomitance with the GSM enhancement, but before introducing link adaptation the older countermeasures will be considered.

- Channel coding introduces redundancy into the data flow increasing its rate by adding information calculated from the source data in order to allow the detection or even the correction of signal errors introduced during transmission. The result of the channel coding is a flow of code words (i.e. blocks as far as concerns block coding). In the case of speech, for example, blocks of 260 bits each are generated every 20 ms at the output of the 13 kbit/s voice encoder. Block coding with parity and convolutional codes, well detailed in **GSM 05.03** introduce redundancy increasing the bits from 260 to 456. Coding schemes make generally use of Puncturing Schemes (PS) acting on block convolutional codes for keeping only q bits out of *pn* through a pre-determined rule. Puncturing permits to reach an efficiency ratio (ratio between the number of useful bits in the source sequence and the number of bits actually transmitted) which is limited to fractions of the form *p/q*, otherwise impossible without puncturing. Parity code adds parity bits to the bits to be convolutionally coded for checking the failure of block convolutional code in error correction. For the sake of completeness a so-called Fire code is prevalently used in fast signalling channel bursts (in-band FACCH) and BCS Header field of GPRS. Fire code is a gender of cyclic code which adds redundancy dedicated to the detection and correction of "bursty" errors. Since a block convolutional code is mainly used for error correction and error often come out in group, the fire code is used in concatenation and noticeably improves the decoded information. Each type of channel has its own characteristic coding scheme. Channel decoding is performed through a de-convolution process which takes advantage from "soft decisions" delivered from the demodulator. Soft decision is an estimated probability of correctness of each detected bit. A convolutional decoder based on the Viterbi algorithm simply exploits Euclidean metrics to implement soft decisions.
- Interleaving consists in mixing up the bits of several code words (code blocks), so that bits which are close to one another in the modulated signal are spread over several code words. Since the error probability of successive bits in the modulated stream is very much correlated, and since channel coding performance is better when errors are de-correlated, interleaving aims at de-correlating errors and their position in code words. In the case of speech, the preceding 456 code bits are reordered and partitioned and diagonal interleaved with 8 timeslot depth to spread the burst errors over more bursts maintaining a reasonable delay of about 37,5 ms (65 burst .periods). De-interleaving is the opposite operation.
- Ciphering modifies the content of a code block through a secret recipe known only by the mobile station and BTS station. The original content (2×57 bit semi-bursts) is encrypted by summing bit by bit to a ciphering flow. Deciphering is the opposite operation. Ciphered coded blocks are differentially encoded before modulation to prevent error propagation.
- Frame alignment takes advantage from Burst formatting which adds some binary information to the ciphered code blocks of 2×57 bit semi-bursts in order to help synchronization and equalization of the received signal and fast signalling. Fig.2 shows that the added information include: 26 bit training sequence, 3+3 TB tail bits, and 1 stealing

flag bit for each 57 bit semi-burst (total 8 bits for the 20 ms speech block) indicating either the semi-burst contains user data or is used in fast associated signalling mode (FACCH). The transmitted training sequence (known to the receiver) has a central peak in its autocorrelation function whose detection from the receiver allows the burst synchronization. Frame alignment is governed by the BTS which measures the TOAs of all the received RF bursts and sends to each mobile station a respective command forcing a delay in the start of transmission in order to maintain constant three frame offset between uplink and downlink bursts.

- Channel equalization usually tempts to reshape the received pulses in order to reduce the intersymbol interference before the demodulation. Contrarily to this definition, an equalizer based on Maximum Likelihood Sequence Estimation (MLSE) criteria, as that based on the Viterbi algorithm, doesn't attempt to equalize the channel in strict sense, but rather uses the knowledge of the channel pulse response (get from the training sequence estimation) to find the data sequence transmitted with the maximum probability. In this area the most recent techniques use beamforming for estimating space and time channel responses. This allows to position the most incoming RF energy towards the directions of the useful signal and its echoes, to the detriment of cochannel interferents. The result is an optimized channel pulse response.
- Block retransmission under Automatic Repeat Request (ARQ) scheme when a code block (different from speech) undergoes one or more residual errors.
- Slow Frequency Hopping (SFH) is a gender of frequency diversity technique descending from the aptitude of Rayleigh fading to be uncorrelated with frequencies spaced sufficiently apart: i.e. 1 MHz. SFH is the interchangeability of the carriers assigned to the physical channels timeslot by timeslot. SFH is carried out inside an orthogonal set of frequencies in use into a cell; the hops are matched between MSs and BTS because of FDD duplexing. For this aim the system refers to a hopping sequence generation algorithm (detailed in **GSM 05.02)** which uses an index MAIO (Mobile Allocation Index Offset) linked to the Frame Number FN.
- Power control (detailed in **GSM 05.08)** is a BSS procedure which step by step modifies, within some range, the uplink/downlink RF transmission power. Power control is based on SACCH Measurement Result message and remedies for path loss and shadow attenuations, further improving spectral efficiency by reducing the overall interference of the system. Secondly it extends battery life of the mobile stations.
- Intra-cell handover (detailed in **GSM 05.08)** is a particular case of the handover procedure charged to switch the mobile station on a free channel of the same cell when transmission quality drops below a given threshold. If an intra-cell handover is successfully the radio link failure can be avoided.

[0023] As already outlined, GPRS service has been added to the GSM in order to achieve higher performance with data handling. The introduction of packet switching capability meets this objective. **TABLE 1 of APPENDIX 1** shows four standard GPRS coding schemes CS-1 to CS-4 relevant to a RLC block. One block of 456 coded bits carries one radio block. CS-1 consists of a half rate convolutional code for FEC and a 40 bit FIRE code for BCS (and optionally FEC). CS-2 and CS-3 are punctured versions of the same half rate convolutional code as CS-1 for FEC. CS-4 has no FEC. Traffic channels exploit CS-1 to CS-4 while signalling channels prefer CS-1. Practical data-rates (kbit/s) achievable on a single GPRS time-slot are shown in the last column of Table 1.

[0024] A subsequent goal of GPRS specifications has been that to increase the data-rate. This aim has been reached by an Enhanced GPRS (EGPRS) version characterized by a higher modulation level, namely 8-PSK (Phase Shift Keying) in combination with additional five coding schemes. In case of 8-PSK modulation a block of 1368 coded bits (456 coded symbols) carries one radio block. While the only GMSK modulation allows to the GPRS users a theoretical bit-rate spanning between 9 and 150 kbit/s (the higher bit-rate being obtained with the poor coding scheme CS-4 and all the eight available time-slots), the 8-PSK modulation allows to the EGPRS users a theoretical bit-rate until 450 kbit/s, triplicating the previous one. In the new EGPRS context, because of the choice between two type of modulations, namely GMSK and 8-PSK, an assignment message shall specify both Modulation and Code type assigned to the channel. Nine combinations of Modulation and Coding Schemes, MCS-1 to MCS-9, are foreseen and detailed in **GSM 05.03, GSM 05.04,** and **GSM 04.60.**

[0025] **TABLE 2 of APPENDIX 1** shows: code rate, data rate, number of coding bits, etc. concerning EGPRS MCS-1 to MCS-9 schemes. In TABLE 2 the column HCS means Header Check Sequence, while the column Family will be explained later. New EGPRS service thanks to the nine MCSi combinations offers several more opportunities for packet data link adaptation. From **TABLE 2** it can be observed that for each type of modulation the grater the code-rate, the greater is the data-rate, because code-rate represents the ratio between the number of useful bits in the source sequence and the number of coded bits. Considering burst having fixed length it results that the higher the code-rate, the poorest is the protection against errors. Higher level modulations (like 8-PSK) are more sensible than lower level modulations (like GMSK) to the causes of RF link degradation and similarly higher code-rates in comparison with lower code rates. Greater sensibility also means faster worsening of the signal delivered to the users as the quality of the RF link worsen. Nevertheless the enhanced opportunity to select one out several combinations of modulation and coding schemes (MCS), enables the system to switch among the various MCSs during run-time to combat the variability

of the RF channel. Link adaptation is just this behavior! TABLE 2 doesn't limit the present invention which is valid also in presence of different high level modulations variously combined with the same or different coding schemes.

[0026] Link adaptation oriented to voice services promotes speech quality compatibly with the variable conditions of the RF link; on the contrary link adaptation oriented to packet data services promotes higher throughputs. In both cases a compromise between data-rate and quality of transmission shall be inevitably pursued when selecting a new modulation and coding scheme. Quality of transmission, and more in general quality of service, plays a main part in a radiomobile system which normally attempts to optimize its own operation by constantly monitoring a lot of parameters. For this aim a variety of measures directed to uplink and downlink transmissions are usually performed, such as (with reference to the single MS): delay of synchronization, channel pulse response, power level of the modulated carrier, power level of the interferent signals, carrier to interferences power ratio (C/I), signal to noise power ratio (S/N), Bit Error Rate (BER), Bit Error Probability (BEP), etc. Incoming useful and interferent signals from the neighbor cells are even monitored to compile a candidate list for handover. The measures performed by the mobile stations are joined to those performed directly by the BTS and sent forward to the BSC to enable its control capability in the opposite direction. The performed measures give support to the most known procedures of the radiomobile system, such as: Cell selection and reselection, Timing advance, Power control, Handover, link adaptation, etc.

[0027] Henceforth packet data transmission will be only considered, because voice/circuit data link adaptation is not particularly relevant for the invention in subject. Consequently the remaining part of the disclosure will be preferably referred to the GPRS/EGPRS improvement of the GSM. Decisions concerning link adaptation for packet data shall be inevitably issued from a high level protocol agent having the supervision of the signalling conveyed through the uplink TBFs and the opportunity to send command through downlink TBFs. The PCU functional block of fig.1 represents a unit charged to manage RLC/MAC blocks and consequently take high level decision about link adaptation. Two different modes of operation are foreseen in the RLC/MAC protocol: acknowledged mode and non-acknowledged mode.

- Acknowledged mode (non-transparent service). Transfer of RLC Data Blocks in the GPRS acknowledged RLC/MAC mode is controlled by a selective ARQ mechanism coupled with the numbering of the RLC Data Blocks participating a Temporary Block Flow. The sending side (the MS or the network) transmits radio blocks within a window and the receiving side sends either Packet Uplink Ack/Nack or Packet Downlink Ack/Nack message when needed. Every such message acknowledges all correctly received RLC Data Blocks up to an indicated block sequence number (BSN), thus "moving" the beginning of the sending window on the sending side. Additionally, a bitmap that starts at the same RLC Data Block is used to selectively request erroneously received RLC Data Blocks for retransmission. The sending side then retransmits the erroneous RLC Data Blocks, eventually resulting in further sliding the sending window. The RLC acknowledged mode shall be used for data applications where the payload content needs to be preserved. It will be the typical mode for Background class (background delivery of e-mails, SMS, download of databases) and Interactive class applications (web browsing). In EGPRS TBF the transfer of RLC Data Blocks in the acknowledged RLC/MAC mode can be controlled by a selective type I ARQ mechanism, or by type II hybrid ARQ mechanism dealing with Incremental Redundancy (IR), both coupled with the numbering of the RLC Data Blocks within one Temporary Block Flow. In the type I ARQ mode, decoding of an RLC Data Block is solely based on the prevailing transmission (i.e. erroneous blocks are not stored). In the type II hybrid ARQ case, erroneous blocks are stored by the receiver and a joint decoding with new transmissions concerning original blocks is done. If the memory for IR operation run out in the MS, the MS shall indicate this by setting an LA/IR bit in the EGPRS PACKET DOWNLINK ACK/NACK message. Type II hybrid ARQ is mandatory in EGPRS MS receivers.
- Non-acknowledged mode (transparent service). The transfer of RLC Data Blocks in the unacknowledged RLC/MAC mode is controlled by the numbering of the RLC Data Blocks participating one Temporary Block Flow, but it does not include any retransmission. The receiving side extracts user data from the received RLC Data Blocks and attempts to preserve the user information length by replacing missing RLC Data Blocks by dummy information bits. Delay sensitive services, such as Conversational class (voice, video conference) and Streaming class applications (one-way real time audio and video) will make use of the RLC unacknowledged mode. The same mechanism and message format for sending temporary acknowledgement messages is used as for acknowledged mode in order to convey the necessary control signalling (e.g. monitoring of channel quality for downlink channel, or timing advance correction for uplink transfers). The sending side (the MS or the network) transmits a number of radio blocks and then polls the receiving side to send an acknowledgement message. A missing acknowledgement message is not critical and a new one can be obtained whenever.

[0028] Quality of Service (QoS), see **GSM 03.60,** takes advantage from both transparent or non-transparent transmissions, as indicated for the services listed above. The two transmission modes differently impact the two QoS classes concerning point-to-point delay and throughput. Unacknowledged packed data is characterized by a fixed point-to-point delay and a variable gross bit-rate, mainly due to the system attempts to maintain a target user bit-rate with the

required quality. On the contrary, due to retransmissions, acknowledged packed data is characterized by a variable point-to-point delay and a variable user bit-rate which can be calculated with the following known expression:

$$\text{Throughput}_{NET} = \text{Throughput}_{MAX}\,(1 - \text{BLER}) \qquad (1)$$

where: Throughput_{NET} is the net user bit-rate; Throughput_{MAX} is the peak user bit-rate; and BLER is the Block Error Rate on the current Modulation and Coding Scheme (MCS).

[0029] Link adaptation is applicable in packet data transmission for both the acknowledged and the unacknowledged transmission modes. Other questions about link adaptation are the following:

- compatibility of the link adaptation with power control;
- effect of frequency hopping on link adaptation;
- the effect of incremental redundancy.

These questions are briefly discussed in the following.

[0030] Both Link Adaptation and Power Control are features that aim at network optimization but, if run independently, may lead to a contrasting situation. Link Adaptation tries to optimize performance (i.e. maximize throughput) for a given radio link quality. This means that if, for instance, radio conditions are improved, the known methods of Link Adaptation try to benefit from this situation and increase the overall throughput by switching to a different (less protected) coding scheme. On the contrary Power Control tries to reduce interference and save power by using the least possible transmit power suitable to achieve a specified C/I ratio (which is consistent with a required minimum performance). In other words PC tends to keep constant the radio link quality thus inhibiting further improvements due to the LA algorithm. Therefore a common strategy has to be decided to make LA and PC work together.

[0031] Frequency hopping increases the variability of the channel so that the choice of an idoneous MCSi shall be conditioned consequently, for example channels having higher variability should require more robust MCSs and consequently lower throughputs.

[0032] Incremental redundancy pertaining to type II hybrid ARQ, differently from type I ARQ, needs a lot of memory to store erroneous block together with multi-bits soft decisions usable in joint decoding the successive retransmitted bits. The overflow probability of an IR buffer de facto increases with the less robust MCSs at the lowest C/I; when this happens lastly stored blocks are discarded and BLER starts to increase. The capacity to contrast worsening of the service clearly depends from the skill of link adaptation to manage this circumstance.

[0033] Now the attention is turned back to the patent application WO 99/12304 filed by ERICSSON whose claim 1 sounds like that: In a communication system, a method for selecting a combination of modulation and channel coding scheme from a plurality of combinations of modulation and channel coding schemes comprising the steps of:

- measuring at least one link quality parameter of an RF link (see claim 2: C/I, BER, received signal level, time dispersion; see claim 8: user data throughput; see claim 9: BLER);
- calculating at least one channel characteristic measure based on the measured at least one link quality parameter (see claim 3: variance; claim 4: mean value);
- estimating user quality values (see claims from 5 to 12) for each one of the combinations of modulation and channel coding schemes based on the calculated channel characteristic measure (variance, mean); and
- selecting a combination of modulation and channel coding schemes (MCSi) on an RF link that provides the best user quality value (see also claim 15: performed during idle states or wait states).

[0034] The step of estimating user quality values is well detailed in the dependent claims as far as it concerns: using of simulation results - using of laboratory results - run time estimating - estimating user data throughput - estimating BLock Error Rate (BLER) - estimating BLER and nominal bit-rate - mapping the calculated media or variance into BLER for each MCSi - estimating speech quality. The above claim doesn't explicitly mention the use of ARQ retransmission. The selected MCSi combination provides for the best user quality value. This claim appears mostly oriented to perform link adaptation for speech service or transparent data service.

[0035] Another independent method claim (16) adds to the preceding claim 1 the feature of: "communicating data using a non-transparent service over an RF link". Contextually user data throughput is estimated instead of quality and the selected MCSi combination provides for the best user data throughput. Clearly this claim appears mostly oriented to perform link adaptation for not-transparent packet data service.

[0036] Another feature added to the independent claims concerns the determination of an optimal transmit power level for each MCSi scheme previously selected by link adaptation. The optimal power is determined based on the

measured C/I and its level is limited by a dynamic range of the power transmitter.

[0037] Main purpose of this prior application is that of calculating the variance of the measured quality parameters, other than the usual averages considered in the oldest methods, for the precise aim to consider the variability of the RF channel when performing a dynamic link adaptation.

COMMENTS ON THIS PRIOR APPLICATION

[0038] Page 25 and Figure 9 of the cited document clarify the wording of the claims.
The clarified method referred to the packet data sounds like that:

1. (see block 112 of fig.9) Measure of some quality parameters (C/I, BER, etc.) and calculation of the relative mean value and variance.
2. (see block 114 of fig.9) Mean value and variance are sent to the inputs of some tables, or mapping functions, obtained from simulations, or through laboratory tests, etc. whose outputs (or mapped values) are the expected BLER(i) for all the MCS(i).
3. (see block 116 of fig.9) By using the know relation: T(i) = T_max(i)*(1-BLER(i)) the throughput T(i) obtainable for that specific condition of mean value and variance is calculated for each MCS(i).
4. (see block 118 of fig.9) In correspondence of the maximum throughput T(i) the respective MCS(i) is selected for coding and modulating the user signal.

[0039] An evident drawback is the intrinsic difficulty of the proposed method, mainly due to the following causes:

1. Time and efforts spent for collecting sufficient measures, further increased to the calculation of averages and variance of the measures (variance needs the knowledge of the average). Measure of C/I is not easy to do.
2. Additional signalling (in the only case of downlink adaptation) to transfer the calculated mean value and variance from the mobile station to the network. Nevertheless this is true only if the measures normally executed for power control and handover were not considered.
3. Off-line calculation of cumbersome tables (or mapping functions) for mapping mean values and variances of a measured parameter into corresponding BLERi for all the MCSi. That because each table of the type BLERi(C/I), or BLERi(BER), etc. shall foreseen entries for two variables and one output for providing BLERi, for example: BLERi(average C/I, variance). So the complete input has reasonably to keep into account several possible variances for each mean value considered in a significant grid.
4. The more complicated the mapping tables are, the more they suffer from sensitivity of the parameters, consequently the empirical representation of a certain BLERi requires to guess exact combinations of mean and variance.
5. Nothing is said on the link adaptation impact in case of non-transparent ARQ with incremental redundancy (type II Hybrid ARQ) implementation. The only use of non-transparent ARQ retransmission without incremental redundancy (type I ARQ) is mentioned in the text. Incremental redundancy impacts the off-line simulations and needs some expedients to be correctly implemented together link adaptation.
6. Optimal transmission power is dependent from the selected MCSi and requires the definition of as many C/I target as the MCSi.

[0040] Being the prior art document silent about point 5, it is not completely proved that the known method works satisfactory with Incremental Redundancy.

OBJECTS OF THE INVENTION

[0041] The main object of the present invention is that to remedy to the defects of the prior art and indicate a method for dynamically optimize data throughput at the radio interfaces of a packet data cellular network.

[0042] Other objects of the invention is that to optimize data throughput at the radio interfaces in presence of slow frequency hopping and/or high user mobility.

[0043] Other objects of the invention is that to optimize data throughput at the radio interfaces in presence of retransmission with incremental redundancy of bad received radio blocks.

[0044] Further objects of the invention is that to harmonize power control and link adaptation mechanisms jointly active at the radio interfaces.

SUMMARY AND ADVANTAGES OF THE INVENTION

[0045] To achieve said objects the subject of the present invention is a method for dynamically optimize data throughput at the radio interfaces of a packet data cellular network, as disclosed in claim 1.

[0046] The link adaptation method of the present invention directly calculates and continuously updates for the active uplink and/or downlink connections, the ratio between the not correctly received blocks and the transmitted blocks, namely the BLER, contrarily to the prior art method in which a BLER is acquired through cumbersome measuring and mapping step

[0047] s. The method of the present invention need not additional signalling for adaptation downlink, that because the number of not correctly decoded blocks constitutes the minimum information that a mobile station have to transmit to the network for requesting retransmission or for quality estimation.

[0048] Once the BLER has been calculated, the problem to be solved is that of how the BLERi for other MCSi would have been for the same channel conditions, in order to establish in base to the known relation (1): $Ti = Ti_{MAX}(1 - BLERi)$ if a change of MCS is needed towards one MCSi having higher throughput Ti. The novelty of the present invention is that relation (1) is not calculated run-time. Instead, using the method of the invention, once BLER has been calculated run-time (without using mapping tables) for the current MCS, a possible change of MCS is established by a simple comparison with two tabulated thresholds. That means to remit to the off-line simulations all the conceptual passages for the determination of the BLER thresholds. The off-line simulations have gone through the conceptual stages that will be detailed later on.

[0049] From above it can be appreciated that the method to perform link adaptation of the invention in subject is quite different from the prior art. In particular it is extremely simpler during run-time execution but also least complex during the off-line simulation devoted to prepare the tabulated thresholds. Obviously the run-time advantage pays the prize of the incapacity to catch the variability of the RF link, if not otherwise provided for; that because the variance is not calculated. The method of the invention provides an effective remedy to the outlined potentially drawback consisting of an additional set of tabulated thresholds for a channel with high variability. The additional set is particularly appreciable in presence of frequency hopping, that because its new thresholds of BLER better fit with the effects of an increased variability of the RF channel. The two sets of tabulated thresholds for channels with or without frequency hopping are both allocated to the BSS (PCU) and either enabled in the occurrence.

[0050] The approach taken for frequency hopping can be extended in line of principle to consider the possible practical RF scenarios. From simulation results it may be noticed that they can be grouped into a few significant cases. For instance, in a typical urban environment, only two different cases can be taken into account: a "low diversity" and a "high diversity" scenario. A first set of thresholds for the "low diversity" scenario should be selected if the cell is characterized by a low user mobility, such as: pico-cells, indoor cells, etc., without Frequency Hopping. A second set of thresholds for the "high diversity" scenario should be selected instead if the cell is characterized by a higher user mobility, such as: ≈50 Km/h mobile speed, or if Frequency Hopping is enabled. The method of the present invention provides the two set of thresholds for "high diversity" and "low diversity" RF channels, in that resolving the problem of the variability of the RF channel.

[0051] The impact of Incremental Redundancy (IR) with Link Adaptation (LA) needs some other considerations out of the mere variability of the RF channel. Some problems arising by combining IR with LA will be outlined in the following, then the solution of these problems by the method of the present invention will be introduced.

[0052] Incremental Redundancy together with link adaptation is known in the art. An exhaustive presentation of the problematic around IR with link adaptation is carried out in the International application number WO 00/49760, also filed by ERICSSON and completely taken into the standard ETSI **GSM 04.60.** A main problem solved by this secondly cited prior art is that of providing suitable overhead signalling messages to enable dynamic changing of the MCS during a connection, taking into account contrasting exigencies between Incremental Redundancy and pure Link Adaptation. A first type of said overhead messages is named LA/IR and corresponds to an additional bit inserted as a flag by the transmitting entity (i.e. the mobile station) in a control word of the RLC control blocks periodically transmitted in uplink to the receiving entity (the network). The LA/IR message provides an explicit request of the preferred operating mode, i.e. either link adaptation or incremental redundancy. This information can then be used by the network when selecting one of two predetermined rules for changing the MCS. For example, if the mobile station MS transmits the LA/IR field with a value which indicates that incremental redundancy is preferred, this implies that it currently has adequate memory capacity to continue to store blocks to support IR combining. This informs the network that the BTS can employ an MCS scheme more aggressive (less robust), taking the link quality estimate report into account. Alternatively, the LA/IR field may instead have a value which indicates that link adaptation is preferred by the mobile station. This may imply that the Mobile station lacks available memory and, therefore, cannot rely on incremental redundancy combining. When the network receives this message may then switch to a second MCS rule makes more conservative (more robust) MCS choices, based on the quality estimates, to ensure that the mobile station achieves sufficient performance without the incremental redundancy combining. Commands to change MCS are enclosed in downlink control blocks.

[0053] A second type of overhead messages of the two mentioned in WO 00/49760 is the value of an additional bit flag named RSEG/NRESEG by means of that the receiving entity informs the transmitting entity whether the MCS for retransmission should be the same or different than the MCS for new blocks transmissions. Before considering the reasons for sending RSEG/NRESEG message a general description of the MCS opportunities for EGPRS is needed. **TABLE 3 of APPENDIX 1** shows that the EGPRS MCS are divided into different families named A ($A_{padding}$), B and C. Each family has a different basic unit of payload: 37 (and 34), 28 and 22 octets respectively. Different code rates within a family are achieved by transmitting a different number of payload units within one Radio Block. For families A and B, 1, 2 or 4 payload units are transmitted, for family C, only 1 or 2 payload units are transmitted. When 4 payload units are transmitted (MCS-7, MCS-8 and MCS-9), these are split into two separate RLC blocks (i.e. with separate sequence numbers and BCSs) within the same Radio Block. These blocks in turn are interleaved over two bursts only, for MCS-8 and MCS-9. For MCS-7, these blocks are interleaved over four bursts. All the other MCSs carry one RLC block interleaved over four bursts. When switching to MCS-3 or MCS-6 from MCS-8, 3 or 6, padding octets, respectively, are added to the data octets. The highlighted structure of the MCSs schemes offers more than one retransmission opportunity to cope with change in the RF channel, for example it's possible under certain restriction, that the message originally pertaining one radio block be retransmitted with more, or less, robust MCS scheme. A change of MCS for the retransmitted message involving a splitting of the payload is said re-segmentation. In case the receiving entity were the network, the downlink control blocks transporting a suitable message include an MCS command which tells the mobile station which MCS should be used for transmitting uplink RLC blocks. The RSEG/NRESEG bit can also be added to the downlink control blocks. In this context a NRSEG asserted (re-segment bit = 0) can be interpreted by the mobile station as meaning retransmissions by the mobile station using the same MCSs as the initial transmissions of RLC blocks; on the other hand a NRSEG negated (re-segment bit = 1) should be interpreted by the mobile station as meaning that blocks to be retransmitted could be re-segmented and transmitted using different MCSs than the initial one. In the latter case, the specific MCS to use for retransmission can be determined by a predetermined rule stored in the receiving entity (mobile station).

[0054] A help in retransmission come from ETSI **GSM 04.60,** paragraph titled "Acknowledged mode operation - Additional functionality in acknowledged EGPRS TBF Mode", in which a procedure is proposed which allows the receiver to operate either in type I or type II hybrid ARQ mode. This procedure says that according to the link quality, an initial MCS is selected for an RLC block. For the retransmissions, the same or another MCS from the same family of MCSs can be selected. E.g. if MCS-7 is selected for the first transmission of an RLC block, any MCS of the family B can be used for the retransmissions. Further, RLC data blocks initially transmitted with MCS4, MCS-5, MCS-6, MCS-7, MCS-8 or MCS-9, can optionally be retransmitted with MCS-1, MCS-2 and MCS-3 respectively, using two radio blocks. In this case, the SPlit Block indicator (SPB) in the header shall be set to indicate that the RLC data block is split, and the order of the two parts. For blocks initially transmitted with MCS-8 which are retransmitted using MCS-6 or MCS-3, padding of the first six octets in the data field shall be applied, and the Coding and Puncturing Scheme (CPS) field shall be set to indicate that this has been done. However, if the transmitter side is the MS and the re-segment bit is not set, the mobile station shall use an MCS within the same family as the initial MCS without splitting the payload for retransmission. The RLC data blocks shall first be sent with one of the initial code rates (i.e., the rate 1/3 encoded data is punctured with the Puncturing Scheme (PS) 1 of the selected MCS). If the RLC Data Block has to be retransmitted, additional coded bits (i.e., the output of the rate 1/3 encoded data which is punctured with PS 2 of the prevailing MCS) shall be sent. If all the codewords (different punctured versions of the encoded data block) have been sent, the procedure shall start over and the first codeword (which is punctured with PS 1) shall be sent followed by PS 2 etc. RLC data blocks which are retransmitted using a new MCS shall at the first transmission after the MCS switch be sent with the puncturing scheme indicated in the **APPENDIX 1 - TABLE 4.** Furthermore, it is mandatory for an EGPRS MS receiver to be able to perform joint decoding among blocks with different MCS's if the combination of MCS's is one of the following:

- MCS-5 and MCS-7,
- MCS-6 and MCS-9.

[0055] The long explanation of the LA/IR technique has twofold meaning, a first one attends to clarify enough this complex argument at the advantage of the disclosure, a second one is that to highlight the lack of indication in the prior art useful to understand the influence of the IR mechanism on the decisional thresholds of BLER. The only reasonable inference on LA/IR from the teaching of the prior art is that IR take over pure LA in case of retransmission with infinite memory pad, but considering the more realistic case of memory saturation, LA is also activated to avoid frequent retransmission. There is a sort of pronounced antagonism between LA and IR at the lower C/I, and the higher BLER values are involved consequently. As the invention in subject is based on the recurs to particular BLER thresholds to maximize the throughput, an important question is that to take realistic thresholds in presence of IR. The prediction of LA/IR interactions is not an easy task at all, because, beyond the probabilistic nature of the phenomenon, the knowledge

of the precise memory size is also required. Memory size at the mobile station side depends on the customer preferences about costs and dimensions of the apparatuses and can't be planned by BSS producer consequently.

[0056] The present invention solves the outlined technical problem starting from the introduction of a variable IR_status which gives continuously updated information to the receiving entity (either the network or the mobile station) about the efficiency of Incremental Redundancy, as disclosed in a relative dependent claim. The evaluation of IR_status is quite simple. Filtered values assumed by the variable IR_status are taken to update the BLER thresholds consequently. Updating is performed by a linear interpolation between two extreme conditions, namely: BLER thresholds relative to lack of IR and BLER thresholds relative to perfect IR. Intermediate and more realistic conditions, so as the two extreme ones, are automatically managed through the updated threshold mechanism. The outlined contrasting behavior between LA and IR, since now remarkable source of problems in the determination of the best adaptation strategy, is no more a problem with the method of the invention extended to the Incremental Redundancy.

[0057] Last argument of the invention is a modified Power Control algorithm having a different goal then the traditional one. The modified algorithm attempts to maintain a C/I_{target} target value for the duration of the whole TBF. The C/I_{target} target is associated to a Peak Throughput per timeslot decided as "Target performance". The association is performed through a curve that represents the maximum achievable Throughput versus C/I. This curve belongs to those simulated off-line during the preliminary step of the Link Adaptation subject of the present invention, in particular to a set having care of the incremental redundancy. Although the upper goal of Power Control, Link Adaptation continues to adapt to radio conditions, switching from one MCS to another, in order to optimize performance on net throughput. This may happen due to the fact that the power control cannot be "perfect" and therefore the actual C/I ratio may be different from the target one. From above it can be argued that the Modified Power Control algorithm complete the Link Adaptation of the present invention working in synergy with it; in that resolving the outlined controversy of the traditional Power Control. Besides, contrarily to the power control of the first cited document of the prior art, it need not separate optimization for each available MCSs.

[0058] From all the above considerations the following substantial advantages of the proposed invention emerge, namely:

- link adaptation runs independently on quality measures, however performed on the ongoing RF signal for traditional Power Control and Handover procedures;
- the variability of the RF channel is neutralized in advance in the adaptation;
- the memory size for Incremental Redundancy is managed in a transparent way;
- power control pursues same goal as link adaptation.

BRIEF DESCRIPTION OF THE DRAWINGS

[0059] Further objects and advantages of the present invention will be made clear by the following detailed description of an embodiment thereof and the annexed drawings given for purely non-limiting explanatory purposes and wherein:

- **fig. 1** shows an GSM and GPRS radiomobile network operating in conformity with the method of the present invention;
- **fig. 2** shows a Time Division Multiple Access (TDMA) multiframe structure common to the GSM and GPRS of fig.1;
- **fig. 3a** shows a GSM traffic channel multiframe;
- **fig. 3b** shows an GPRS traffic channel multiframe;
- **fig. 4** shows the mapping of higher level frames into radio blocks belonging to the EGPRS multiframe of fig.3b;
- **fig. 5** shows the functional blocks of a mobile station MS/UE of fig.1 operating in conformity with the method of the present invention;
- **fig.6** shows the functional blocks of a base station BTS of fig.1 operating in conformity with the method of the present invention;
- **figures 7, 8, 9, 10, 11, 12, 13, 14** show graphic representations of simulation results used to implement the method of the present invention;
- **figures 15 and 16** show respective flow charts of the link adaptation method of the present invention;
- **fig.17** shows a graphic representation used to implement the power control function at the Um interface of fig.1.
- **APPENDIX 1 - TABLE 1** includes coding parameters for GPRS coding schemes.
- **APPENDIX 1 - TABLE 2** includes coding parameters for EGPRS modulation and coding schemes.
- **APPENDIX 1 - TABLE 3** represents payload families used in the EGPRS coding schemes.
- **APPENDIX 1 - TABLE 4** includes Puncturing Schemes for EGPRS.
- **APPENDIX 1 - TABLE 5** includes modulation and coding schemes to be used for retransmissions when re-segmentation is not enabled.
- **APPENDIX 1 - TABLE 6** includes modulation and coding schemes to be used for retransmissions when re-seg-

mentation is enabled.

DETAILED DESCRIPTION OF AN EMBODIMENT OF THE INVENTION

[0060] The arguments of **Figures 1, 2, 3a, 3b and 4,** so as **TABLES 1, 2, 3, and 4** of **APPENDIX 1** have already been duly discussed above in the text.

[0061] **Figure 5** shows a block diagram of a mobile station MS/UE suitable to implement the present invention in conjunction with the BSS subsystem of fig.1. The mobile station MS/UE includes a Transmitting Section and a Receiving Section both controlled through a Control Processor that further controls a Frequency Synthesizer & Hopping unit common to the two sections. A Duplexer filter conveys to the antenna the RF output signals of the Transmitting section and to the input of the Receiving section the RF signal received on the antenna. For the sake of simplicity an oscillator and a TDMA timing generator are not shown'in fig.5. The Transmitting Section includes the following functional blocks: Input devices, Speech coder, Channel coder, Interleaver, Ciphering, Burst formatter, GMSK / 8-PSK Modulator, BB/IF/ RF UP converter, and RF Power amplifier. Input devices include a microphone with relative A/D converter and a Key-board & Adapter. The Receiving Section in its turn includes the following functional blocks: Image filter, RF amplifier, RF/IF/BB Down converter, LEV, CH filter, A/D converter, Correlator and MLSE (Viterbi) estimator, Burst disassembler, Deciphering, De-Interleaver, Channel decoder, Speech decoder & Voice amplifier, Output devices (Earphone, PC Monitor, Fixed Disk, etc.). In conformity with its A, B, or C operative class, the mobile station MS/UE is able to operate with both voice and data input devices, simultaneously or not. Class A users have one time slot allocated for speech and one ore more others to the EGPRS service. Dual considerations apply to the Output devices. As already mentioned the present invention is prevalently addressed to packet data, so the blocks Input devices and Output devices will exemplify known data terminals for inputting or outputting data respectively. Those terminals include pads and adapter circuits for synchronizing, storing, adapting format and rate of the incoming/outgoing digital blocks. Considering the Transmitting section at first, Channel coder accepts data from Input devices and provides a relevant EGPRS coding scheme, selected from those reproduced on TABLES 1 and 2. For this aim a CPS-TX-SEL signal is outputted from the Control Processor. Channel coder provides for: block code, parity code, convolutional code and fire code; it further accepts and codifies DATA-INS signalling RLC blocks (such as measures) from the Control Processor. Coded blocks are sent to the cascade of Interleaver, Ciphering, and Burst formatter to perform the relative digital treatments as explained in the introduction. A formatted burst is delivered to the GSM / 8PSK Modulator which starts performing a differential encoding followed by either a GMSK or 8-PSK modulation. Control Processor selects the modulation type by sending to the Modulator a MOD-TX-SEL signal, always in respect of the MCS schemes listed in TABLE 2. The base band analog modulated signal is firstly translated to IF frequency and then to RF frequency by means of suitable up conversion mixers; each conversion stage is followed by a band pass filtering stage. The RF transmission signal reaches the input of a variable gain Power amplifier whose output is coupled to a transmission port of a Duplexer filter coupled to the Mobile station antenna. The downlink RF signal coming from the BTS reaches the Mobile station antenna and leaves a receiving port of the Duplexer filter, crosses an Image filter and reaches the input of a Reception low noise amplifier whose output is connected to a frequency down converter. The down conversion is carried out by two cascaded stages: a first one converts from RF to IF, and the second one from IF to base band BB. The second stage also splits the converted signal into the in-phase I and in-quadrature Q components. The base band I, Q components are filtered by two channel filters CH matched to the transmitted pulse and then analog-to-digital converted. The two copies of the digitalized reception burst arrive at the two inputs of a Correlator/Synchronizer, acting like a matched filter to the training sequence, which extracts the correlation peak for detecting the initial instant of the transmission. The same correlative process also estimates the pulse response of the channel supplied to an MLSE estimator based on the Viterbi algorithm. This algorithm acts on a sequentially built-up trellis having as many nodes (reiterated at each symbol time T) as the states $S = M^L$ of the receiver, corresponding to all the possible combinations generated from M words (symbols)) of a modulation alphabet over L symbol times (where L is the significant length of the initially estimated channel pulse response). Starting from a known initial state, the progressive path along the trellis will depend on the effective transmitted sequence. All the possible transmitted sequences are distinguished each other through a respective path metric which constitutes the Likelihood function to be gradually maximized by accumulating transition metrics. At every new symbol time M transition metrics Δ are calculated in correspondence of the M branches departing from each preceding node to reach a number M of successive nodes. A transition metric (or branch metric) is the Euclidean distance between the level of the received symbol and the level that should have been received in correspondence of a supposed transition on the trellis. Among all the branches departing from a node only a survivor one is selected to prolong a trellis path passing through that node, namely that having the maximum actual path metric. So doing, a drastic cut of the complexity is performed because the original number of states is maintained at each step. Among all the survived paths at the time T the candidate sequence is the one which has the maximum path metric. Going back along the trellis for a certain number of steps it can be appreciate that only a path survives, which is associated to a segment of the transmitted sequence. More precision is obtained delaying the decision of the MLSE estimator until

the end of the burst. At the output of the MLSE estimator a copy of the original burst is reproduced and each bit is accompanied with three bits soft decisions indicating its received level. The estimated burst is delivered to a cascade of the following blocks: Burst disassembler, Deciphering, De-interleaver, and Channel decoder; the last carries out the specified operations in respect of TABLES 1 and 2 by exploiting soft decisions. Control Processor generates the following two signals: MOD-RX-SEL and CPS-RX-SEL towards MLSE estimator and Channel decoder respectively. That because modulation and/or code scheme of the received signal can differ from the transmitted ones. MLSE estimator operates with either GMSK or 8-PSK modulation, obviously with different trellis and branch metric expression. Channel decoder uses Soft decisions to carry out convolutional decoding and also takes advantage from the mentioned Incremental redundancy strategy supported by an Incremental Redundancy buffer for temporarily storing RLC blocks to be retransmitted under ARQ. A buffer overflow activates a signal IRout directed to the Control Processor. Decoded RLC signalling blocks, indicated with DATA-EXTR, are extracted and sent to the Control Processor for the correct interpretation and execution (such as: Power control, Timing Advance, Handover, etc.). Channel decoder detects and counts errors before error correction and informs the Control Processor by sending a signal BER having the usual meaning of Bit Error Rate. Since decoding is good an OK Flag is set. Decoded RLC blocks concerning traffic are sent to the appropriate output devices in conformity with the selected A, B or C user class. Control Processor governs the main operational procedures of the Mobile station MS/UE through a first and a second group of signals indicated as TRANSMITTING SECTION control and RECEIVING SECTION control, respectively directed to the two sections. Among these signals the following three are pointed out: MAIO, RSSI and PC. MAIO is directed to the Frequency Synthesizer & Hopping unit in order to provide indication for frequency hopping and handover. Signal RSSI is generated from a circuit LEV which samples, A/D converts, and measures the strength of the received signal, and noise during idle. Control Processor block includes a memory RAM for temporarily store Level 2 and Level 3 signalling messages.

[0062] **Figure 6** shows a block diagram of a Base Transceiver Station (BTS) suitable to implement the present invention in conjunction with the BSS subsystem of fig.1 and the Mobile Stations MS/UE of fig.5. The mobile BTS includes a Transmitting Section and a Receiving Section both controlled through a BTS Control Processor that further controls a Frequency Synthesizer & Hopping unit common to the two sections. The two Sections and the BTS Control Processor are connected to an A-Bis INTERFACE functional block for receiving/outputting one or more PCM link at 2 Mb/s or PCU frames incoming from or outgoing to the BSC (fig.1). A Duplexer filter conveys to the antenna the RF output signals of the Transmitting section and to the input of the Receiving section the RF signal received on the antenna. For the sake of simplicity a clock generator/extractor and a TDMA timing generator are not shown in fig.6. The Transmitting Section includes the following functional blocks: Base band processing 1...n, GMSK or 8-PSK digital modulators 1....n, MULTICARRIER DIGITAL TRANSMITTER. The Receiving Section includes the following functional blocks: Base band processor 1,...,n, Equalizer & Demodulator 1,....,n, MULTICARRIER DIGITAL RECEIVER, and an Image filter. Starting from the Transmitting Section, the A-bis INTERFACE block extracts from the PCM link or PCU frames all the n elementary fluxes concerning CH1....CHn channels relevant to the n users. CH1....CHn fluxes reach respective Base band processors to undergo all the digital treatments as: coding (parity, convolutional fire), interleaving, ciphering, burst formatting, and differential coding. Convolutional coding provides a relevant EGPRS coding scheme, selected from those reproduced on TABLES 1 and 2. The n coded signals outputted from the Base band processors reach as many GMSK/8-PSK digital modulators to be digitally modulated in respect of the MCS schemes listed in TABLE 2. The n modulated digital signals reach as many DUCs (Digital Up Converters) inside the MULTICARRIER DIGITAL TRANSMITTER. Each DUC further receives a respective local oscillator signal $f_{IF\text{-}DUC}$ for the translation of its base band input signal to a prefixed position inside the overall Intermediate Frequency band. For this aim the $f_{IF\text{-}DUC}$ signals are digital sinusoids. The n IF digital signals are summed up by a digital adder working at the higher $f_{IF\text{-}DUC}$ frequency, and the multicarrier IF resulting signal is D/A converted and wide band filtered before reaching the input of an IF/RF mixer piloted by a $f_{OL\text{-}TX}$ local oscillator signal to the up conversion at radiofrequency. The RF signal at the output of the mixer is sent to an RF power amplifier. The output of the RF power amplifier is connected to the TX port of the Duplexer filter, while the RX port is connected to the Image filter placed at the input of the MULTICARRIER DIGITAL RECEIVER. The RF filtered signal is amplified and down converted to IF by an RF/IF mixer piloted by a $f_{OL\text{-}RX}$ local oscillator signal. The multicarrier analog IF signal is anti-alias filtered and fed to the input of n DDCs (Digital Down Converters) inside the MULTICARRIER DIGITAL RECEIVER. Each DDC further receives a respective local oscillator signal $f_{IF\text{-}DDC}$ for the translation to base band its input signal relevant to a prefixed position inside the overall Intermediate Frequency band. For this aim the $f_{IF\text{-}DDC}$ signals are digital sinusoids. The n digital base band signals reach as many Equalizer & Demodulator to be demodulated in respect of the MCS schemes listed in TABLE 2. The same arguments as the Viterbi's estimator of fig.5 are still valid. The demodulated signals are sent to the Base band processors to undergoes: Burst disassembling, Deciphering, De-interleaving, and Channel decoding in respect of TABLES 1 and 2. Finally the decoded data relevant to CH1,...CHn channels are delivered to the A-bis INTERFACE functional block to be assembled into one 2 M/bit outgoing PCM link or PCU frames. Control Processor block includes a memory RAM for temporarily store Level 2 and Level 3 signalling messages for all the n users. The BTS Control Processor governs the main operational procedures of the Mobile station MS/UE through a first and group of signals indicated as "TRANSMITTING

SECTION control", "Signalling insertion"; and a second group of signals indicated as "RECEIVING SECTION control", "Signalling extraction". Among these signals a MAIO group is directed to the Frequency Synthesizer & Hopping unit in order to provide indication for frequency hopping and handover as far as concerns all the DUC and DDC circuits. Control Processor block includes a memory RAM for temporarily store Level 2 and Level 3 signalling messages for all the n users. Extracted signalling concerns, for example: measures transmitted uplink by all the Mobile stations (level, BER, C/I, OK flag, etc.), the statuses of the IRout overflow indicators, etc. Inserted signalling concerns, for example: Power control commands directed to each Base band processor, Timing advance commands, selection of the individual MCS scheme for transmission and/or reception, etc.

[0063] The reference frame of a known GSM-EGPRS system has been completed at this point of the disclosure. So the basis for the introduction of the features typical of the invention are given. The relevant means of the invention to carry out uplink and/or downlink TBF link adaptation constitute a particular combination of known and new means like the following list, in which when not expressly mentioned they are preferably allocated to the PCU and either confined in the firmware or in dedicated circuits:

- memory matrix tables for memorizing as many sets of digital values intended as BLER thresholds; the tables being managed by the Packet Control Unit (PCU). The thresholds being calculated off-line in a way that will be soon illustrate and they are valid both for uplink and downlink adaptation;
- means of the PCU for the selection of the tables;
- means allocated both to the BTS and the mobile stations for decoding RLC received blocks, optionally capable of joint decoding Incremental Redundancy bits;
- means allocated both to the BTS and the mobile stations for detecting and storing RLC blocks erroneously received;
- means allocated both to the BTS and the mobile stations for retransmitting erroneously received blocks;
- means for calculating BLER of an active TBF by filtering a variable indicating the RLC blocks not correctly received;
- means for checking the performance of the Incremental Redundancy detection;
- means for filtering a variable indicating the effectiveness of the Incremental Redundancy detection;
- means for continuously updating the BLER thresholds on the basis of said effectiveness variable;
- means to compare the calculated BLER with the updated BLER thresholds in order to obtain a criterion for changing the actual MCS;
- means of the PCU to command a new MCS on the basis of said criterion for changing the actual MCS;
- means of the BSC for updating the transmission power level of each uplink/downlink channel in order to maintain a fixed target throughput independently on the MCSs.

[0064] With reference to the **Figures 7 to 14** the preliminary off-line simulation step useful for determining the various sets of BLER thresholds is now considered. Those Figures are to be considered two at a time, such as: Fig.7 and 8; 9 and 10; 11 and 12; 13 and 14. The arguments relative to the first couple of Figures 7 and 8 are generally still valid for the other couple of figures. **Fig. 7** shows some curves of net throughput (kbit/s) in function of C/I (dB) for several Modulation and Coding Schemes. **Fig. 8** shows correspondent curves of BLER (dB or %) in function of C/I (dB) for the same MCSs of fig.7. Four MCSs are represented in fig.7 indicated with a, b, c, d; they respectively coincide with MCS1, MCS3, MCS6, and MCS9 of **TABLE 2.** It can be appreciate that the listed MCSs is a subset of all the possible MCSs constituting a sequence of MCSs arranged by increasing nominal throughputs. Curves of **fig.7** are referred to a standard channel TU3 (Typical Urban - 3 ray model) without Frequency Hopping and without Incremental Redundancy (only Type I ARQ is admitted), they are valid for both uplink and downlink TBFs. The depicted values are the result of a computer simulation refined and validated through on field measures. Curves of **fig.7** are derived from curves of **fig. 8** by using the relation (1).

[0065] Because of the trends of the various MCS curves of fig.7 are not similar to that of parallel lines, six different cross-points are visible in correspondence of as many values of C/I. Cross-points are characterized by equal net throughputs for at least two MCS curves. Cross-points relevant for the present invention are only the three relative to adjacent MCSs in the ordered sequence, namely: a-b, b-c, and c-d. In order to maximize throughput the higher order MCS should be selected at the right of the switching point, while the lower order MCS should be chosen when the RF channel conditions are at the left of the cross point. This behavior is due to the decreasing protection of the higher MCS at the lower C/I and the consequent retransmission of the errored radio blocks. Referring to the previous cross-points of fig.7 the 'ideal' switching points between two adjacent MCSi could be the following:

MCS a ↔ MCS b:	C/I ≈ 1.5 dB
MCS b ↔ MCS c:	C/I ≈ 7.5 dB
MCS c ↔ MCS d:	C/I ≈ 16 dB

[0066] But C/I values are difficult to estimate in a real network, while other parameters, such as BLER, can be calculated directly. The Link Adaptation algorithm here proposed will then be based on direct BLER measurements. The previous calculated 'ideal' C/I switching points now correspond to the following 'ideal' couples of BLER thresholds mapped on the curves of **fig.8:**

$$\text{MCS a} \leftrightarrow \text{MCS b: C/I} = 1.5\ \text{dB} \Rightarrow BLER_{MCS1 \rightarrow MCS3} = Tab,\ BLER_{MCS3 \rightarrow MCS1} = Tba;$$

$$\text{MCS b} \leftrightarrow \text{MCS c: C/I} = 7.5\ \text{dB} \Rightarrow BLER_{MCS3 \rightarrow MCS6} = Tbc,\ BLER_{MCS6 \rightarrow MCS3} = Tcb;$$

$$\text{MCS c} \leftrightarrow \text{MCS d: C/I} = 16\ \text{dB} \Rightarrow BLER_{MCS6 \rightarrow MCS9} = Tcd,\ BLER_{MCS9 \rightarrow MCS6} = Tdc.$$

[0067] Net throughput is then maximized changing the MCS according to these BLER threshold values. If actual BLER falls below the upgrade threshold (Tab, Tbc, Tcd) the algorithm switches to the next (less protected) available MCS. If actual BLER instead exceeds the downgrade threshold (Tab, Tbc, Tcd) the algorithm switches to the previous (more protected) available MCS. For example, if BLER goes below Tbc, while using MCS b, then a change to MCS c will be decided. On the contrary, if BLER goes above Tba, while using MCS b, then a change to MCS a will be decided.

[0068] If the RF environment changes, the MCS's performances curves change as well. Therefore the 'ideal' switching points depend on the actual RF environment. As an example, 'ideal' switching points may be different if Frequency Hopping is enabled or disabled in the network. Though the possible RF scenarios are virtually infinite, as already anticipated in the introduction, in a typical urban environment, only two different cases can be taken into account: a "low diversity" and a "high diversity" scenario.

[0069] The "low diversity" scenario corresponds to the family of curves represented in **Figures 7 and 8** and should be selected if the cell is characterized by a low user mobility, such as: pico-cells, indoor cells, etc. without Frequency Hopping.

[0070] The "high diversity" scenario corresponds to the family of curves represented in **Figures 9 and 10** and should be selected if the cell is characterized by a higher user mobility, such as ≈50 Km/h mobile speed, or if Frequency Hopping is enabled. Simulation results represented in Figs. 9 and 10 have been obtained in absence of IR.

[0071] For each specific RF scenario different upgrade switching points and downgrade switching points are derived through simulations and on field measures. These values of the switching points constitute as many sets of thresholds stored in matrix tables. Once the particular RF scenario has been assigned, the corresponding matrix table is selected, containing all the ideal switching points (downgrade/upgrade switching points from/to all MCSs) for that case. The initial MCS has to be defined as said later on:

[0072] Things are further complicated when type II Hybrid ARQ (Incremental Redundancy) is utilized. In the Figures 11, 12 and 13, 14 simulation results with IR (and infinite memory) are presented for the same scenarios described above. More precisely, simulation results represented in **Figures 11 and 12** concern cells characterized by "low diversity" in presence of Incremental Redundancy. In this case it can be seen that MCS d outperforms all others MCS for a wide range of C/I ratios and the setting of the switching points will require some further considerations. Simulation results represented in **Figures 13 and 14** concern cells characterized by "high diversity" in presence of Incremental Redundancy. Even here further considerations are necessaries. In any case it should be noticed that again, even in presence of Incremental Redundancy, the resulting performance depends on the actual RF scenario. Moreover results depend on the amount of memory available for Incremental Redundancy. Anyway, as a result, when IR is taken into account, different BLER threshold values should be considered. Even these values should be stored in matrix tables, one for each possible RF scenario.

[0073] With reference to the **Figures 15 and 16,** the Link Adaptation method subject of the present invention is discussed. For the sake of simplicity the method is like a flow-chart of a program which controls a microprocessor inside the PCU (fig.1). In the reality the various steps of the program interact with the involved protocol procedures and signalling. The previous off-line step for obtaining the BLER threshold matrix tables shall be considered as a preliminary part of the method. **Fig.15** concerns a simplified method valid for a packed data scenario without Incremental Redundancy and either characterized by low or high variability. **Fig.16** differs from fig.15 in that Incremental Redundancy is considered. Matrix tables relative to the **Figures 8, 10, 12,** and **14** have been respectively indicated as **Table A, B, C,** and **D.**

[0074] The method of **fig.15** starts with step **S1** which addresses the TBFs adaptation either uplink or downlink. Presently uplink TBFs are considered, successively the modifications for downlink TBFs will be introduced. In the subsequent step **S2** the connection is established and the Initial Modulation and Coding Scheme is decided. The initial MCS will be set by default, unless some other information is available. In step **S3** at the network side value of BLER

is continuously updated, at each received radio block, by checking if RLC blocks have been carefully received or not. BLER at instant n, for a given TBF connection, is obtained by a digital filter having a pulse response exponentially decreasing with time discrete n as indicated by the following law:

$$BLER_n = f_1(BLER_{n-1})+f_2(s_n) \tag{2u}$$

where:

- n is the iteration index spanning one radio block period of 20 ms;
- $s_n = 0$ if the RLC block at instant n has been correctly received (and the MCS is the "commanded MCS");
- $s_n = 1$ if the RLC block at instant n has not been correctly received;
- $$s_n = \frac{1}{K}\sum_{k=1}^{K} s_{n,k} \tag{3u}$$

  if more than one RLC block is received s_n is the average of the values calculated for single blocks. De facto more than one RLC block for a given TBF can be received at the same time instant n, due to 1) multislot allocation, 2) MCSs supporting two RLC blocks at a time.
- $f_1(BLER_{n-1})$ is a first weight function of the preceding filtered BLER value relative to the "commanded MCS" (i.e. actual MCS) blocks only, taking values inside the interval 0 - 1;
- $f_2(s_n)$ is a second weight function of the variable s_n, taking values inside the interval 0 - 1;

[0075] Taking into consideration the teaching of standard ETSI **GSM 05.08** about time filtering of quality variables, for analogy, expression (2u) now assumes the following expression:

$$BLER_n = (1-\beta\cdot\frac{X_n}{R_n})\cdot BLER_{n-1}+\beta\cdot\frac{X_n}{R_n}\cdot s_n \tag{2u'}$$

where:

- n is the iteration index spanning one radio block period of 20 ms;
- x_n is equal to 1 if "at least" one RLC block for the considered TBF with the "commanded MCS" is received at time instant n, otherwise is set to 0;
- s_n has been already defined;
- β is the forgetting factor:

$$\beta = 1/T_{AVG},$$

[0076] T_{AVG} being the filtering period in multiples of a radio block;

- R_n denotes the reliability of the n_{th} BLER measurement and is expressed as follows:

$$R_n = (1-\beta)\cdot R_{n-1}+\beta\cdot x_n; \qquad R_{-1}= 0 \tag{4u}$$

  R_n is the output of a running average filter that helps to keep track of the reliability of the filtered BLER measurements. In fact R_n is used in (2u') to decide the weight between the new measurement (s_n) and the old measurements ($BLER_{n-1}$). Looking at the formulas, it comes out that at time instants where no measurement exists (no RLC blocks are received for the considered TBF), $BLER_n$ will not be updated. On the contrary, when a measurement exists, $BLER_n$ will be updated weighting new and old contributions, so to obtain the desired exponentially decreasing (with discrete time n) filter impulse response. The reliability filter is initialized at the beginning of a transmission (n=0) setting $R_{-1} = 0$.

By the comparison of expression (2u) with (2u') it results that:

$$f_1(BLER_{n-1}) = (1-\beta\cdot\frac{X_n}{R_n})\cdot BLER_{n-1}$$

$$f_2(S_n) = \beta\cdot\frac{X_n}{R_n}\cdot s_n.$$

Besides the two weight functions $f_1(BLER_{n-1})$ and $f_2(s_n)$ have balanced weights, so that an arbitrary weight increasing of $f_2(s_n)$ also involves an equal weight decreasing of $f_1(BLER_{n-1})$, and vice versa.

[0077] In next step **S4** the presence of Frequency Hopping is checked. If the answer in step **S4** is negative, the case of low diversity environment is checked in the successive step **S5.** Affirmative answer in step **S4** enters step **S6** in which the BLER filtered at step **S3** is compared to the upgrade and downgrade thresholds stored in **Table A.** Negative answer in step **S4** enters step **S6'** where the filtered BLER is compared to the thresholds stored in **Table B.** The comparison using **Table B** is also performed if Frequency Hopping were found active in the preceding step **S4.** Thresholds could be generalized in this way: put MCSx the actual MCS, MCSy the next available less protected one, and MCSz the previous available more protected one, then the appropriate thresholds will be:

$$\text{Upgrade thresholds } (UP\_th_n): \quad BLER_{MCSx \rightarrow MCSy}$$

$$\text{Downgrade thresholds } (DN\_th_n): \quad BLER_{MCSx \rightarrow MCSz}.$$

[0078] Reaching the step **S7** either from **S6** or **S6',** the occurrence is checked of an MCS switching in consequence of the previous comparisons. If in step **S7** the actual value of BLER doesn't cross any thresholds the subsequent step **S8** performs an unitary increment of index n, then in step **S9** the active state of the actual TBF is monitored. Until TBF is active the respective BLER is continuously monitored from the cycle of steps **S3 - S10** to check the conditions for switching from the actual MCS; if during the cycle the TBF elapses the incoming step **S10** resets BLER and R and the program waits for another TBF. If during the cycle **S3 - S10** the actual BLER falls below the value UP_th_n, then MCSx is switched to MCSy in step **S11.** Alternatively, if during the cycle **S3 - S10** the actual BLER exceeds the value DN_th_n, MCSx is switched to MCSz in step **S11.** When commanding the new MCS to the MS, in a PACKET UPLINK ACK/NACK or PACKET TIMESLOT RECONFIGURE message, the PCU can also set the re-segment bit to the proper value. In general, for retransmissions, setting the re-segment bit to '1' requires the mobile station MS/UE to use an MCS within the same family as the initial transmission and the payload may be split. Instead setting the re-segment bit to '0' requires the mobile station shall use an MCS within the same family as the initial transmission without splitting the payload. **TABLES 5 and 6 of APPENDIX 1** show MCS schemes to use for retransmission after switching to a different MCS. **TABLE 5** is valid for re-segment bit = 1, while **TABLE 6 is valid** for re-segment bit = '0'. According to the invention, in the case under description (no Incremental Redundancy mode), the re-segment bit is always set to "1". Whenever the Modulation and Coding Scheme is changed, BLER and R variable are set to zero in the successive step **S12** and the filtering process is re-started from step **S3.**

[0079] Additional advantages of the disclosed method are mostly due to the filtering step **S3,** they are:

- Considering that at each iteration index n used in digital filter (2u) and (2u') (20 ms) could not correspond an RLC block for the intended TBF, due to the MAC scheduling mechanism, and that, on the contrary, a constant BLER filtering window is preferable in expression (2u'), then the reliability filter (4u) provides the way to keep constant the "actual" BLER filtering window, in that independent on the number of TBFs multiplexed on the same TS. Consequently the BLER digital filter (2u') is taken back to the RLC blocks effectively received in order to maintain the right exponentially decreasing impulse response.
- Only blocks encoded with the present MCS contribute to BLER calculation. In other words, retransmissions with a different MCS don't have any impact on BLER calculation for the actual MCS.

[0080] With reference to the **fig.16** changes in respect to the **fig.15** are now discussed to be introduced in the link adaptation method due to the Incremental Redundancy. The first five steps **S1** to **S5** are the same as those of fig.15, in particular the filtering step **S3.** Additional problems arise when the actual thresholds for the BLER comparison shall be determined. These problems are of different nature and must be checked consequently, so step **S6 (S6')** is delib-

erately introduced for this aim. During this step the following routine is executed to set the logic value of a variable IR_check that contributes to the issue of an IR_status variable which gives information about the efficiency of Incremental Redundancy at the BTS:

IF

{

there has been an header error (this implies that IR for the expected block(s) is useless),

OR

if memory for IR is exhausted (no IR is possible for the expected block(s)),

OR

if soft decisions could not have been stored due to any other reason (again, no IR for the expected block(s))

}

THEN IR at time instant n is considered as "not working", $IR\_check_n=0$

ELSE IR at time instant n is considered as "working", $IR\_check_n=1$.

[0081] The IR_status is then filtered in step **S7 (S7')** using the same approach used for BLER in step **S3;** in particular using a digital filter having a response exponentially decreasing with discrete time n as indicated by the following law:

$$IR\_status_n = f_1(IR\_status_{n-1})+f_2(IR\_check_n) \tag{5u}$$

were function f_1 and f_2 follow the same laws as used in the BLER calculation. The analogy is extended to the most detailed function:

$$IR\_status_n = (1-\beta\cdot\frac{X_n}{R_n})\cdot IR\_status_{n-1}+ \beta\cdot\frac{X_n}{R_n}\cdot IR\_check_n \tag{6u}$$

where: x_n, R_n, and β are the same values used in the BLER calculation.

[0082] Differently from the preceding method of fig.15, BLER thresholds stored in the matrix tables are not immediately usable, since such thresholds depend on the used MCS but on the IR efficiency as well. So the successive step **S8(S8')** is charged to calculate suitable thresholds for taking IR into account. The new thresholds are the result of a linear interpolation between two extreme cases, namely: perfect IR (IR_status = 1), and IR totally lacking (IR_status = 0). Each case making reference to its own matrix tables. Absence of IR needs tables **A** and **B,** while perfect IR needs tables **C** and **D,** besides tables **A** and **C** simulate low diversity channels respectively without and with IR, while tables **B** and **D** high diversity channels respectively without and with IR. Consequently the linear interpolation in step **S8** taking care of low diversity channels recurs to tables **A** and **C,** while the linear interpolation in step **S8'** taking care of high diversity channels recurs to tables **B** and **D.**

[0083] Indicating with $BLER_{MCSx\_wIR \to MCSy\_wIR}$, and $BLER_{MCSx\_wIR \to MCSz\_wIR}$ respectively the new upgrade UP_th_n and downgrade DN_th_n thresholds for perfect IR, the linear interpolations calculated either in step **S8** or **S8'** assume the following expressions:

$$UP\_th_n = (1 - IR\_status_n) \times BLER_{MCSx \to MCSy} + IR\_status_n \times BLER_{MCSx\_wIR \to MCSy\_wIR} \tag{7u}$$

$$DN\_th_n = (1 - IR\_status_n) \times BLER_{MCSx \rightarrow MCSz} + IR\_status_n \times BLER_{MCSx\_wIR \rightarrow MCSz\_wIR} \quad (8u)$$

[0084] The successive step **S9** is charged to compare BLER filtered in step **S3** with the new thresholds (7u) and (8u) either coming from step **S8** or **S8'**, then in step **S10** the occurrence of an MCS switching in consequence of the previous comparisons is checked. If from the check of step **S10** it results that in step **S9** the actual BLER doesn't cross any UP_th_n or DN_th_n thresholds, the subsequent step **S11** performs an unitary increment of index n, then in step **S12** the active state of the actual TBF is monitored. Until TBF is active the respective BLER is continuously monitored from the cycle of steps **S3 - S12** to check the conditions for switching from the actual MCS; if during the cycle the TBF elapses the incoming step **S13** resets BLER and R variables and the program waits for another TBF. If during cycle **S3 - S12** the actual BLER falls below the value UP_th_n, then in step **S14** MCSx is switched to MCSy. Alternatively, if during the cycle **S3 - S12** the actual BLER exceeds the value DN_th_n, in step **S14** MCSx is switched to MCSz. When commanding the new MCS to the mobile station, in a PACKET UPLINK ACK/NACK or PACKET TIMESLOT RECONFIGURE message, the PCU unit can also set the re-segment bit to the proper value. If $IR\_status_n < 0.5$ then IR is considered as "not-properly working" and the re-segment bit is set to '1'. On the contrary, if $IR\_status_n > 0.5$ then IR is considered as "properly working" and the re-segment bit is set to '0'. For retransmissions the previous considerations are still valid so as **TABLES 5 and 6** of **APPENDIX 1.**

[0085] Whenever the Modulation and Coding Scheme is changed, in the successive step **S15** BLER and R variables are set to zero and the filtering process is re-started from step **S3.**

[0086] Additional advantage of the disclosed method is that it is independent on the memory size at the BTS. In fact if there is so much memory as the IR_status variable will always be close to 1, then in step **S9** the "perfect IR" thresholds $BLER_{MCSx\_wIR \rightarrow MCSy\_wIR}$ and $BLER_{MCSx\_wIR \rightarrow MCSz\_wIR}$ will always be used, because they are prevailing in expressions (7u) and (8u). On the contrary, if the BTS has as low memory as IR_status variable will always be close to 0, then in step **S9** the "no IR" thresholds $BLER_{MCSx \rightarrow MCSy}$, and $BLER_{MCSx \rightarrow MCSz}$ will always be used, because they are prevailing in expressions (7u) and (8u). It can be appreciated that through expressions 7u) and (8u) a sort of automatic switch between the two extreme conditions is performed.

[0087] The disclosure of how performing uplink adaptation with Incremental Redundancy carried out with reference to the **Fig.16** (the most general case), is nearly completely applicable to the downlink adaptation. Downlink adaptation is carried out by the network (BTS, BSC, PCU), as well as for uplink adaptation, but in case of downlink adaptation the receiving entities are the mobile stations which have to transmit to the network their own surveys on block decoding and the residual state of the IR memory. In practice, once the connection is established, BLER is updated at the PCU with the information provided by the EGPRS PACKET DOWNLINK ACK/NACK message, reported by the MS upon periodic request (polling) from the network. The exploitation by the PCU of the polled information suitable for calculating BLER imposes to change the time iteration index n used in the expressions (2u) and (2u') of the digital filters, and in the other descending expressions. In downlink case, time iteration index n for a given TBF connection must be replaced with reporting instant k for the same connection. So, the most general expression (2u) becomes:

$$BLER_k = f_1(BLER_{k-1}) + f_2(s_k) \quad (2d)$$

while more detailed expression (2u') requires a modification of the two weights and of the reliability variable R (expression 4u) to consider the greater lasting effect of reporting instant k. In that the following expression is valid for downlink adaptation:

$$BLER_k = \left(1 - \frac{\beta}{R_k}\right) \cdot BLER_{k-1} + \frac{\beta}{R_k} s_k \quad (2d')$$

where:

- k is the reporting instant lasting m RLC blocks;
- $$s_k = \frac{Nack\_blocks}{Sent\_blocks}$$

Nack_blocks: number of badly received RLC blocks among those sent with the present MCS.

Sent_blocks: number of blocks sent with the present MCS in the previous polling period.

- β is the forgetting factor as already defined;
- R_k denotes the reliability of the filtered BLER measurement expressed as in the following:

$$R_k = (1-\beta)^m \cdot R_{k-1} + \beta; \qquad R_{-1}=0 \tag{4d}$$

where m is the number of radio blocks that elapsed since the last EGPRS PACKET DOWNLINK ACK/NACK message was received at the PCU. Again, R_k is the output of a running average filter that helps to keep track of the reliability of the filtered BLER measurements. In fact R_k is used to decide the weight between the new measurement (s_k) and the old measurements ($BLER_{k-1}$). When a new measurement exists (an EGPRS PACKET DOWNLINK ACK/NACK message is received), $BLER_k$ will be updated weighting new and old contributions, so to obtain the desired exponentially decreasing (with discrete time n) filter impulse response. The reliability filter is initialized at the beginning of a transmission (k=0) setting $R_{-1} = 0$. Differently from expression (4u) that uses iteration index n, expression (4d) uses iteration index k spanning several time index n, nevertheless the two expression shall perform comparable filtering function on the same filtering window, exponent m used in expression (4d) provides to this task by increasing the effect of the single iteration k in a way to opportunely dampen the old measure and reinforce the new input as if m consecutive RLC blocks were filtered in the meanwhile.

[0088] Considerations about the Incremental Redundancy, to say expression (5u) and (6u) both pertaining to IR_status and IR_check variables remain formally unchanged by using the reporting instant k. The same applies to the settlement of upgrade and downgrade thresholds through the expressions (7u) and (8u). In particular, when an EGPRS PACKET DOWNLINK ACK/NACK message is received, the MS_OUT_OF_MEMORY bit is checked:

IF

{

this bit is set (no more memory for IR is available at the MS)

}

THEN IR at instant k is considered as "not working", $IR\_check_k=0$

ELSE IR at instant k is considered as "working", $IR\_check_k=1$.

[0089] The IR status is then filtered using the same approach used for BLER:

$$IR\_status_k = f_1(IR\_status_{k-1}) + f_2(IR\_check_k) \tag{5d}$$

were function f_1 and f_2 follow the same laws as used in the BLER calculation. The analogy is extended to the most detailed function:

$$IR\_status_k = \left(1-\frac{\beta}{R_k}\right)\cdot IR\_status_{k-1} + \frac{\beta}{R_k}\cdot IR\_check_k \tag{6d}$$

where R_k (4d) and β have already been introduced.

[0090] The IR_status variable gives information about the efficiency of Incremental Redundancy at the MS.

[0091] The linear interpolations for updating all the upgrade and downgrade tabulated BLER thresholds associated to each available MCS take now the following expressions:

$$UP\_th_k = (1 - IR\_status_k) \times BLER_{MCSx \rightarrow MCSy} + IR\_status_k \times BLER_{MCSx\_wIR \rightarrow MCSy\_wIR} \tag{7d}$$

$$DN\_th_k = (1 - IR\_status_k) \times BLER_{MCSx \rightarrow MCSz} + IR\_status_k \times BLER_{MCSx\_wIR \rightarrow MCSz\_wIR} \tag{8d}$$

where: UP_th_k and DN_th_k are the upgrade and downgrade thresholds respectively.

[0092] Additional advantages of downlink adaptation method are still those listed for uplink.

[0093] With reference to **fig.17** a modified Power Control algorithm for pursuing aims as pursued by Link Adaptation object of the present invention is now disclosed. Without limiting the invention the modified Power Control algorithm attempts to maintain a high data throughput of transmitting entities subjected to Link Adaptation with Incremental Redundancy. The modified Power Control takes part in the off-line preliminary step of link adaptation by making use of the simulation curves of net throughput (kbit/s) in function of C/I (dB) for several Modulation and Coding Schemes. The curve that grants the maximum achievable throughput (i.e. the envelope of all the curves corresponding to the different MCS in the Incremental Redundancy case) is used and reproduced in **Fig. 17** Target can be derived from the Peak Throughput QoS class requested by the mobile station. Let T_P be the Peak Throughput, then a Peak Throughput per timeslot, indicated as T_{PxTS}, is calculated:

$$T_{PxTS} = T_P / N_{TS} \tag{9}$$

where N_{TS} is the number of timeslots allocated to the TBF; i.e. N_{TS} is the minimum between the number of allocable timeslots and the number of timeslots that can be handled by the MS due to its multislot class.

[0094] Once T_{PxTS} is set on the ordinate axis of the curve "Maximum achievable throughput", the curve itself associates to the T_{PxTS} point a target C/I_{target} value on the abscissa axis. In other words the couple of points (C/I_{target}, T_{PxTS}) is marked on the "Maximum achievable throughput" curve. C/I_{target} target value constitutes the goal of the modified Power Control algorithm. Traditional Power Control algorithm attempts to minimize transmission power compatibly with a minimum fixed quality of the transmitted signal checked by the receiving entity. To reach this aim it needs to handle measures included in channel quality reports carried by associated control channels. Once the measures have been acquired, the traditional Power Control algorithm starts to increase, or decrease, step by step the transmitted power until the outlined goal on minimum quality has been checked back from the measures. Modified Power Control algorithm works as the traditional one but with a different goal, namely it tries to maintain the C/I_{target} target value for the duration of the whole TBF. The Link Adaptation algorithm subject of the present invention, on the other hand, continues to adapt to radio conditions, switching from one MCS to another, in order to optimize performance on net throughput. This may happen due to the fact that the power control cannot be "perfect" and therefore the actual C/I ratio may be different from the target one. From above it can be argued that the Modified Power Control algorithm works in synergy with the link adaptation, in that resolving the controversy outlined in the prior art.

APPENDIX 1

TABLE 1: Coding parameters for the GPRS coding schemes

Scheme	Code rate	USF	Pre-coded USF	Radio Block excl. USF and BCS	BCS	Tail	Coded bits	Punctured bits	Data rate kb/s
CS-1	1/2	3	3	181	40	4	456	0	9.05
CS-2	≈2/3	3	6	268	16	4	588	132	13.4
CS-3	≈3/4	3	6	312	16	4	676	220	15.6
CS-4	1	3	12	428	16	-	456	-	21.4

TABLE 2: Coding parameters for the EGPRS coding schemes

<table>
<tr><th>Scheme</th><th>Code rate</th><th>Header Code rate</th><th>Modulation</th><th>RLC blocks per Radio Block (20ms)</th><th>Raw Data within one Radio Block</th><th>Family</th><th>BCS</th><th>Tail payload</th><th>HCS</th><th>Data rate kb/s</th></tr>
<tr><td>MCS-9</td><td>1.0</td><td>0.36</td><td rowspan="5">8PSK</td><td>2</td><td>2x592</td><td>A</td><td rowspan="3">2x12</td><td rowspan="3">2x6</td><td rowspan="9">8</td><td>59.2</td></tr>
<tr><td>MCS-8</td><td>0.92</td><td>0.36</td><td>2</td><td>2x544</td><td>A</td><td>54.4</td></tr>
<tr><td>MCS-7</td><td>0.76</td><td>0.36</td><td>2</td><td>2x448</td><td>B</td><td>44.8</td></tr>
<tr><td>MCS-6</td><td>0.49</td><td>1/3</td><td>1</td><td>592
544+48</td><td>A</td><td rowspan="6">12</td><td rowspan="6">6</td><td>29.6
27.2</td></tr>
<tr><td>MCS-5</td><td>0.37</td><td>1/3</td><td>1</td><td>448</td><td>B</td><td>22.4</td></tr>
<tr><td>MCS-4</td><td>1.0</td><td>0.53</td><td rowspan="4">GMSK</td><td>1</td><td>352</td><td>C</td><td>17.6</td></tr>
<tr><td>MCS-3</td><td>0.80</td><td>0.53</td><td>1</td><td>296
272+24</td><td>A</td><td>14.8
13.6</td></tr>
<tr><td>MCS-2</td><td>0.66</td><td>0.53</td><td>1</td><td>224</td><td>B</td><td>11.2</td></tr>
<tr><td>MCS-1</td><td>0.53</td><td>0.53</td><td>1</td><td>176</td><td>C</td><td>8.8</td></tr>
<tr><td colspan="11">NOTE: the italic captions indicate the padding.</td></tr>
</table>

TABLE 4 – Puncturing Schemes (PS)

MCS switched from	MCS switched to	PS of last transmission before MCS switch	PS of first transmission after MCS switch
MCS-9	MCS-6	PS 1 or PS 3	PS 1
		PS 2	PS 2
MCS-6	MCS-9	PS 1	PS 3
		PS 2	PS 2
MCS-7	MCS-5	any	PS 1
MCS-5	MCS-7	any	PS 2
all other combinations		any	PS 1

APPENDIX 1

TABLE 3 - MODULATION AND CODING SCHEMES FOR EGPRS

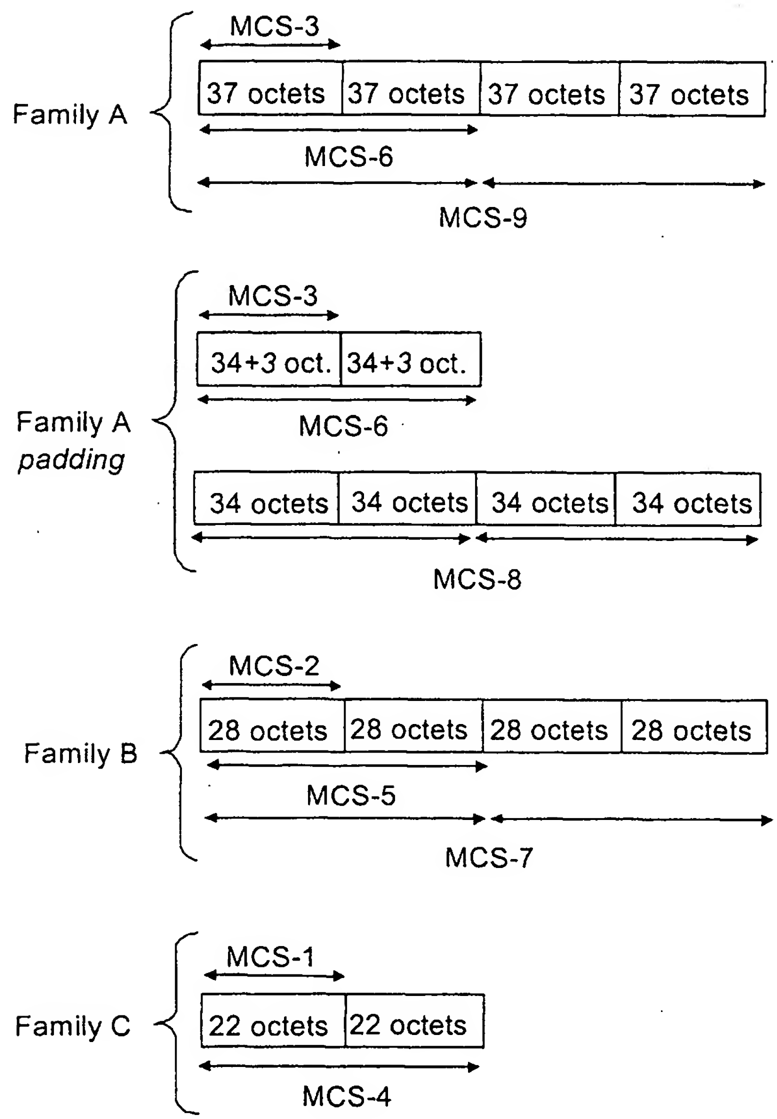


TABLE 5 - MCS to use for retransmissions when re-segmentation (re-segment bit set to '1') is carried out (specified as a function of the scheme used for the initial transmission)

Scheme used for initial transmission	Scheme to use for retransmissions after switching to a different MCS										
	MCS-9 Commanded	**MCS-8 Commanded**	**MCS-7 Commanded**	**MCS-6-9 Commanded**	**MCS-6 Commanded**	**MCS-5-7 Commanded**	**MCS-5 Commanded**	**MCS-4 Commanded**	**MCS-3 Commanded**	**MCS-2 Commanded**	**MCS-1 Commanded**
MCS-9	MCS-9	MCS-6	MCS-6	MCS-6	MCS-6	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3
MCS-8	MCS-8	MCS-8	MCS-6 (pad)	MCS-6 (pad)	MCS-6 (pad)	MCS-3 (pad)	MCS-3 (pad)	MCS-3 (pad)	MCS-3 (pad)	MCS-3 pad)	MCS-3 (pad)
MCS-7	MCS-7	MCS-7	MCS-7	MCS-5	MCS-5	MCS-5	MCS-5	MCS-2	MCS-2	MCS-2	MCS-2
MCS-6	MCS-9	MCS-6	MCS-6	MCS-9	MCS-6	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3
MCS-5	MCS-7	MCS-7	MCS-7	MCS-5	MCS-5	MCS-7	MCS-5	MCS-2	MCS-2	MCS-2	MCS-2
MCS-4	MCS-4	MCS-4	MCS-4	MCS-4	MCS-4	MCS-4	MCS-4	MCS-4	MCS-1	MCS-1	MCS-1
MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3
MCS-2	MCS-2	MCS-2	MCS-2	MCS-2	MCS-2	MCS-2	MCS-2	MCS-2	MCS-2	MCS-2	MCS-2
MCS-1	MCS-1	MCS-1	MCS-1	MCS-1	MCS-1	MCS-1	MCS-1	MCS-1	MCS-1	MCS-1	MCS-1

TABLE 6 - MCS to use for retransmissions when re-segmentation is not (re-segment bit set to '0') allowed specified as a function of the scheme used for the initial transmission)

Scheme used for initial transmission	Scheme to use for retransmissions after switching to a different MCS										
	MCS-9 Commanded	**MCS-8 Commanded**	**MCS-7 Commanded**	**MCS-6-9 Commanded**	**MCS-6 Commanded**	**MCS-5-7 Commanded**	**MCS-5 Commanded**	**MCS-4 Commanded**	**MCS-3 Commanded**	**MCS-2 Commanded**	**MCS-1 Commanded**
MCS-9	MCS-9	MCS-6	MCS-6	MCS-6	MCS-6	MCS-6	MCS-6	MCS-6	MCS-6	MCS-6	MCS-6
MCS-8	MCS-8	MCS-8	MCS-6 (pad)	MCS-6 (pad)	MCS-6 (pad)	MCS-6 (pad)	MCS-6 (pad)	MCS-6 (pad)	MCS-6 (pad)	MCS-6 (pad)	MCS-6 (pad)
MCS-7	MCS-7	MCS-7	MCS-7	MCS-5	MCS-5	MCS-5	MCS-5	MCS-5	MCS-5	MCS-5	MCS-5
MCS-6	MCS-9	MCS-6	MCS-6	MCS-9	MCS-6	MCS-6	MCS-6	MCS-6	MCS-6	MCS-6	MCS-6
MCS-5	MCS-7	MCS-7	MCS-7	MCS-5	MCS-5	MCS-7	MCS-5	MCS-5	MCS-5	MCS-5	MCS-5
MCS-4	MCS-4	MCS-4	MCS-4	MCS-4	MCS-4	MCS-4	MCS-4	MCS-4	MCS-4	MCS-4	MCS-4
MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3	MCS-3
MCS-2	MCS-2	MCS-2	MCS-2	MCS-2	MCS-2	MCS-2	MCS-2	MCS-2	MCS-2	MCS-2	MCS-2
MCS-1	MCS-1	MCS-1	MCS-1	MCS-1	MCS-1	MCS-1	MCS-1	MCS-1	MCS-1	MCS-1	MCS-1

Claims

1. Method for dynamically optimize data throughput at the radio interfaces of a packet data cellular network having at disposal of said interfaces one or more type of modulations and a plurality of coding schemes for providing several combinations of modulation-and-coding schemes, or MCSs, more or less protected against transmission errors and usable for transmitting bursts of data, packed-up in blocks, between mobile stations (MS) and the serving base station (BTS), and vice versa, the throughput optimization being dynamically pursued selecting MCSs capable to grant the highest net throughput when the quality of the RF link changes starting from an initially assigned one, the selection exploiting empirical representations of the net data throughput allowable by each MCS in function of a quality parameter of the RF link, like the carrier to interference ratio, **characterized in that** it includes:

   - an off-line preliminary step for obtaining from said empirical representations some tabulated values (A, B, C, D) of Block Error Rate, or BLER, to be used like upgrade and/or downgrade thresholds associated to each available MCS for determining as many switching points between MCS having the immediately lower or higher error protection; and

   the following steps cyclically repeated for all the duration of a temporary connection set up with said initial MCS:

   - updating at each new incoming block of data an averaged value of BLER evaluated in correspondence of a commanded MCS presently in use;
   - comparison of said averaged BLER of the actual MCS with the associated upgrade and/or downgrade thresholds;
   - changing the actual MCS into said MCS immediately less error protected when the averaged BLER is lower than said upgrade threshold; or
   - changing the actual MCS into said MCS immediately more error protected when the averaged BLER is higher than said downgrade threshold.

2. Method for dynamically optimizing data throughput according to claim 1, **characterized in that** said averaged value of BLER is obtained by weighting both the preceding values of BLER and the actual decisions on errored blocks, using a digital filter having a pulse response exponentially decreasing with discrete time n spanning a block period.

3. Method for dynamically optimizing data throughput according to claim 2, **characterized in that** said pulse response of BLER digital filter is obtained by summing up two weight functions both accepting samples with the "commanded MCS", a first one to weigh the preceding values of BLER and a second one to weigh the actual decisions on errored blocks.

4. Method for dynamically optimizing data throughput according to claim 3, **characterized in that** said first and second weight functions have balanced weights, so that an arbitrary increasing of the weight of the first function also involves an equal decreasing of the weight of the second function, and vice versa.

5. M'ethod for dynamically optimizing data throughput according to claim 4, **characterized in that** the weight of said first and second weight functions are both equally varied in order to compensate the missing filtering effect of possible lacking blocks, **in that** making the outlined pulse response possible.

6. Method for dynamically optimizing data throughput according to claim 5, **characterized in that** the variation of said weights are carried out by making the said first and second weight functions further depending on a reliability function which tracks the age of the received blocks.

7. Method for dynamically optimizing data throughput according to any claim from 3 to 5, **characterized in that** said temporary connection is dedicated to transfer packet data from a selected mobile station to the base station, and said pulse response of BLER digital filter is obtained by means of the following function:

$$BLER_n = f_1(BLER_{n-1})+f_2(s_n)$$

   where:

- n is the iteration index spanning one block period;
- $s_n = 0$ if the block at instant n has been correctly received;
- $s_n = 1$ if the block at instant n has not been correctly received;
-
$$s_n = \frac{1}{K}\sum_{k=1}^{K} s_{n,k}$$
if K blocks are received for the considered connection;
- $f_1(BLER_{n-1})$ is said first weight function, taking values inside the interval 0 - 1;
- $f_2(s_n)$ is said second weight function of the variable s_n relative to the decision on the errored blocks, taking values inside the interval 0 - 1;

8. Method for dynamically optimizing data throughput according to claim 6, **characterized in that** said first and second weight functions assume the following expressions:

$$f_1(BLER_{n-1}) = (1-\beta\cdot\frac{x_n}{R_n})\cdot BLER_{n-1}$$

$$f_2(s_n) = \beta\cdot\frac{x_n}{R_n}\cdot s_n$$

where:

- x_n is equal to 1 if "at least" one RLC block for the considered connection with the commanded MCS is received at time instant n, otherwise is set to 0;
- $\beta = 1/T_{AVG}$ is a forgetting factor and T_{AVG} being the filtering period in multiples of a radio block;
- $R_n = (1-\beta)\cdot R_{n-1} + \beta\cdot x_n$; $R_{-1} = 0$ is said reliability function.

9. Method for dynamically optimizing data throughput according to any claim from 3 to 6, **characterized in that** said temporary connection is dedicated to transfer packet data from the base station to a selected mobile station, and said pulse response of BLER digital filter is obtained by means of the following function:

$$BLER_k = f_1(BLER_{k-1}) + f_2(s_k)$$

where:

- k is the reporting instant lasting m blocks;
-
$$s_k = \frac{Nack\_blocks}{Sent\_blocks}$$

Nack_blocks: number of badly received blocks among those sent with the present MCS;

Sent_blocks: number of blocks sent with the present MCS in the previous polling period:

- $f_1(BLER_{k-1})$ is said first weight function, taking values inside the interval 0 - 1;
- $f_2(s_k)$ is said second weight function of the variable s_k relative to the decision on the errored blocks, taking values inside the interval 0 - 1.

10. Method for dynamically optimizing data throughput according to claim 8, **characterized in that** said first and second weight functions assume the following expressions:

$$f_1(BLER_{k-1}) = (1-\frac{\beta}{R_k})\cdot BLER_{k-1}$$

$$f_2(s_k) = \frac{\beta}{R_k} \cdot s_k$$

where:

- $\beta = 1/T_{AVG}$ is a forgetting factor and T_{AVG} being the filtering period in multiples of a radio block;
- $R_k = (1-\beta)^m \cdot R_{k-1} + \beta$; $R_{-1} = 0$ is said reliability function.

11. Method for dynamically optimizing data throughput according to one of the preceding claims, **characterized in that** a condition of buffer full and other main causes making retransmission with incremental redundancy inapplicable, are continuously checked on the ongoing connection at the receiver side and the relevant piece of information whether incremental redundancy is properly working or not is forwarded to the network (PCU).

12. Method for dynamically optimizing data throughput according to one of the preceding claims, **characterized in that** said tabulated values of BLER thresholds (A, B, C, D) are subdivisible into at least two former tables (A, B), a first one (A) for low-diversity RF channels and a second one (B) for high-diversity RF channels, considering as high-diversity a channel **characterized by** frequency hopping or high user mobility, and low-diversity a channel without frequency hopping and with low user mobility.

13. Method for dynamically optimizing data throughput according to claim 12, **characterized in that** each of said former tables of BLER thresholds (A, B) has a companion table (C, D) whose BLER thresholds are set taking further into account the additional effect of the incremental redundancy.

14. Method for dynamically optimizing data throughput according to claim 13, **characterized in that** all said upgrade and/or downgrade tabulated BLER thresholds (A, B, C, D) associated to each available MCS are updated at the network side (PCU) at every reception of the incremental redundancy relevant information, making use of a linear interpolation between a threshold stored into a former table (A, B) and the correspondent threshold stored into the companion table (C, D), the interpolation exploiting a network-provided variable, named for mere convenience IR_status, measuring the averaged status of incremental redundancy for the aim of unbalancing the entity of the interpolation either towards said companion table (C, D) when incremental redundancy prevails, or towards the former table (A, B) on the contrary case.

15. Method for dynamically optimizing data throughput according to claim 14, **characterized in that** said variable IR_status measuring the averaged status of incremental redundancy is obtained by weighting both the preceding values of IR_status and the actual values of a variable, named for mere convenience IR_check, taking value 1 if incremental redundancy is properly working, or value 0 on the contrary, using a digital filter having a pulse response exponentially decreasing with discrete time n spanning a block period.

16. Method for dynamically optimizing data throughput according to claim 15, **characterized in that** said temporary connection is dedicated to transfer packet data from a selected mobile station to the base station, and said pulse response of IR_status digital filter is obtained by means of the following function:

$$IR\_status_n = f_1(IR\_status_{n-1}) + f_2(IR\_check_n)$$

were:

- n is the iteration index spanning one block period;
- f_1 and f_2 are weight functions following the same laws as used in the BLER calculation.

17. Method for dynamically optimizing data throughput according to claim 16, **characterized in that** said first and second weight functions assume the following expressions:

$$f_1(IR\_status_{n-1}) = (1-\beta \cdot \frac{X_n}{R_n}) \cdot IR\_status_{n-1}$$

$$f_2(IR\_check_n) = \beta \cdot \frac{X_n}{R_n} \cdot IR\_check_n$$

where: R_n takes a formal expression as that used in the BLER calculation, while x_n and β are the same.

18. Method for dynamically optimizing data throughput according to claim 16 or 17, **characterized in that** said linear interpolation for updating all said upgrade and/or downgrade tabulated BLER thresholds associated to each available MCS take the following expressions:

$$UP\_th_n = (1 - IR\_status_n) \times BLER_{MCSx \rightarrow MCSy} + IR\_status_n \times BLER_{MCSx\_wIR \rightarrow MCSy\_wIR}$$

$$DN\_th_n = (1 - IR\_status_n) \times BLER_{MCSx \rightarrow MCSz} + IR\_status_n \times BLER_{MCSx\_wIR \rightarrow MCSz\_wIR}$$

where:

- UP_th_n and DN_th_n are said upgrade and downgrade thresholds respectively;
- $BLER_{MCSx \rightarrow MCSy}$ is an upgrade threshold stored into a said former table;
- $BLER_{MCSx\_wIR \rightarrow MCSy\_wIR}$ is an upgrade threshold stored into a said companion table;
- $BLER_{MCSx \rightarrow MCSz}$ is a downgrade threshold stored into a said former table;
- $BLER_{MCSx\_wIR \rightarrow MCSz\_wIR}$ is a downgrade threshold stored into a said companion table.

19. Method for dynamically optimizing data throughput according to claim 15, **characterized in that** said temporary connection is dedicated to transfer packet data from the base station to a selected mobile station, and said pulse response of IR_status digital filter is obtained by means of the following function:

$$IR\_status_k = f_1(IR\_status_{k-1}) + f_2(IR\_check_k)$$

were:

- k is the reporting instant lasting m blocks;
- f_1 and f_2 are weight functions following the same laws as used in the BLER calculation.

20. Method for dynamically optimizing data throughput according to claim 19, **characterized in that** said first and second weight functions assume the following expressions:

$$f_1(IR\_status_{k-1}) = (1 - \frac{\beta}{R_k}) \cdot IR\_check_{k-1}$$

$$f_2(IR\_check_k) = \frac{\beta}{R_k} \cdot IR\_check_k$$

where: R_k takes a formal expression as that used in the BLER calculation, and β is the same.

21. Method for dynamically optimizing data throughput according to claim 19 or 20, **characterized in that** said linear interpolation for updating all said upgrade and/or downgrade tabulated BLER thresholds associated to each available MCS take the following expressions:

$$UP\_th_k = (1 - IR\_status_k) \times BLER_{MCSx \rightarrow MCSy} + IR\_status_k \times BLER_{MCSx\_wIR \rightarrow MCSy\_wIR}$$

$$DN\_th_k = (1 - IR\_status_k) \times BLER_{MCSx \rightarrow MCSz} + IR\_status_k \times BLER_{MCSx\_wIR \rightarrow MCSz\_wIR}$$

where:

- UP_th_k and DN_th_k are said upgrade and downgrade thresholds respectively;
- $BLER_{MCSx \rightarrow MCSy}$ is an upgrade threshold stored into a said former table;
- $BLER_{MCSx\_wIR \rightarrow MCSy\_wIR}$ is an upgrade threshold stored into a said companion table;
- $BLER_{MCSx \rightarrow MCSz}$ is a downgrade threshold stored into a said former table;
- $BLER_{MCSx\_wIR \rightarrow MCSz\_wIR}$ is a downgrade threshold stored into a said companion table.

22. Method for dynamically optimizing data throughput according to one of the preceding claims, **characterized in that** a modified power control works in parallel with the MCS switching link adaptation and the modified power control includes the following steps:

- off-line calculation of the expression:

$$T_{PxTS} = T_P / N_{TS},$$

where: T_{PxTS} is the Peak Throughput per timeslot; T_P is the Peak Throughput derived from the Quality of Service Class of the connection, and N_{TS} is the minimum between the number of allocable timeslots and the number of timeslots that can be handled by the MS due to its multislot class;

- off-line mapping of the calculated T_{PxTS} on a simulated curve depicting the maximum achievable net throughput in function of the values of Carrier versus Interference C/I, and obtaining from the curve a target C/I_{target} value;
- exploiting the C/I_{target} for all the duration of the ongoing connection as a goal to be maintained by the network (BSC, BTS) exploiting the Power and Interference measures at the receiver side.

GSM (DCS) - GPRS (Enhanced) SYSTEM

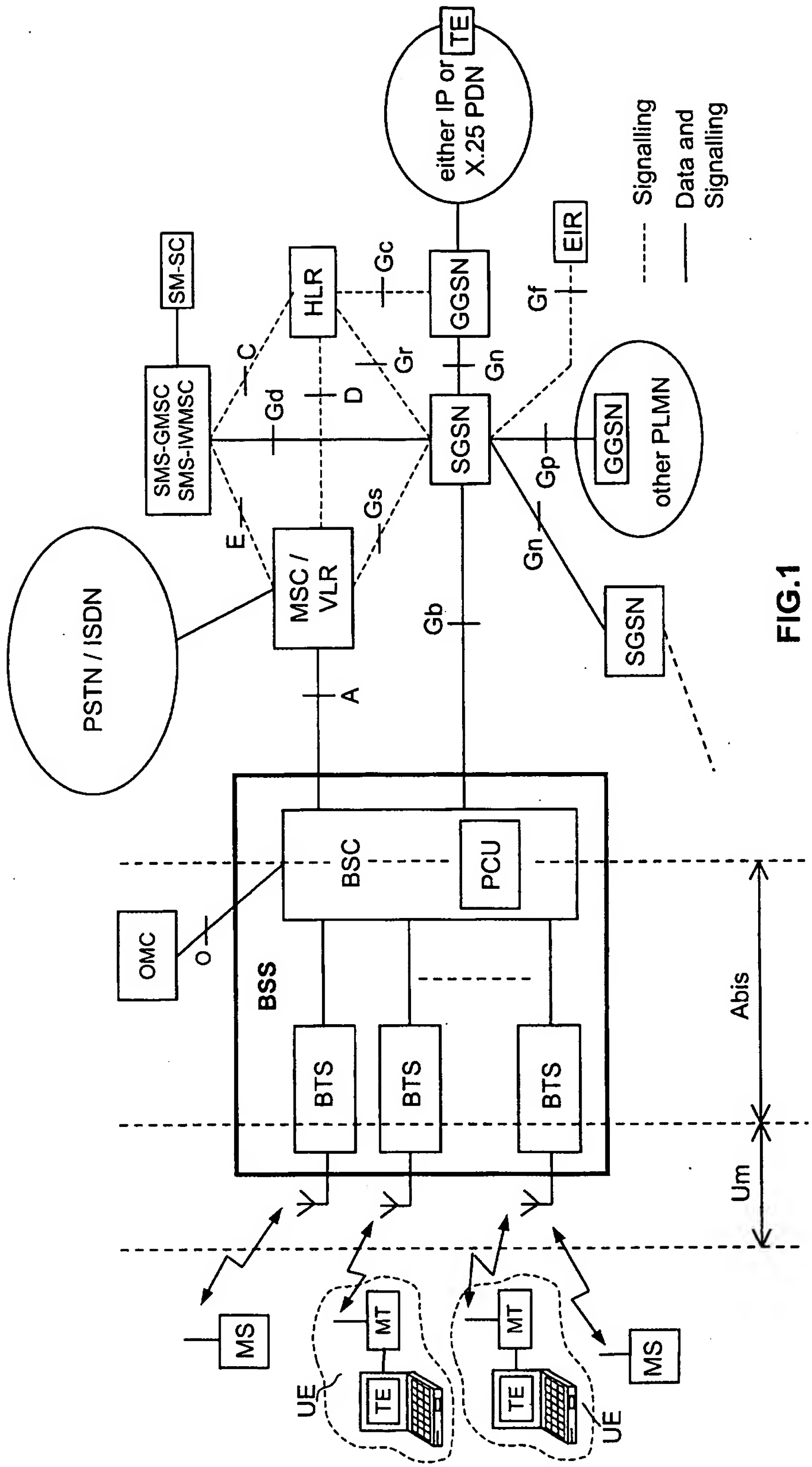


FIG.1

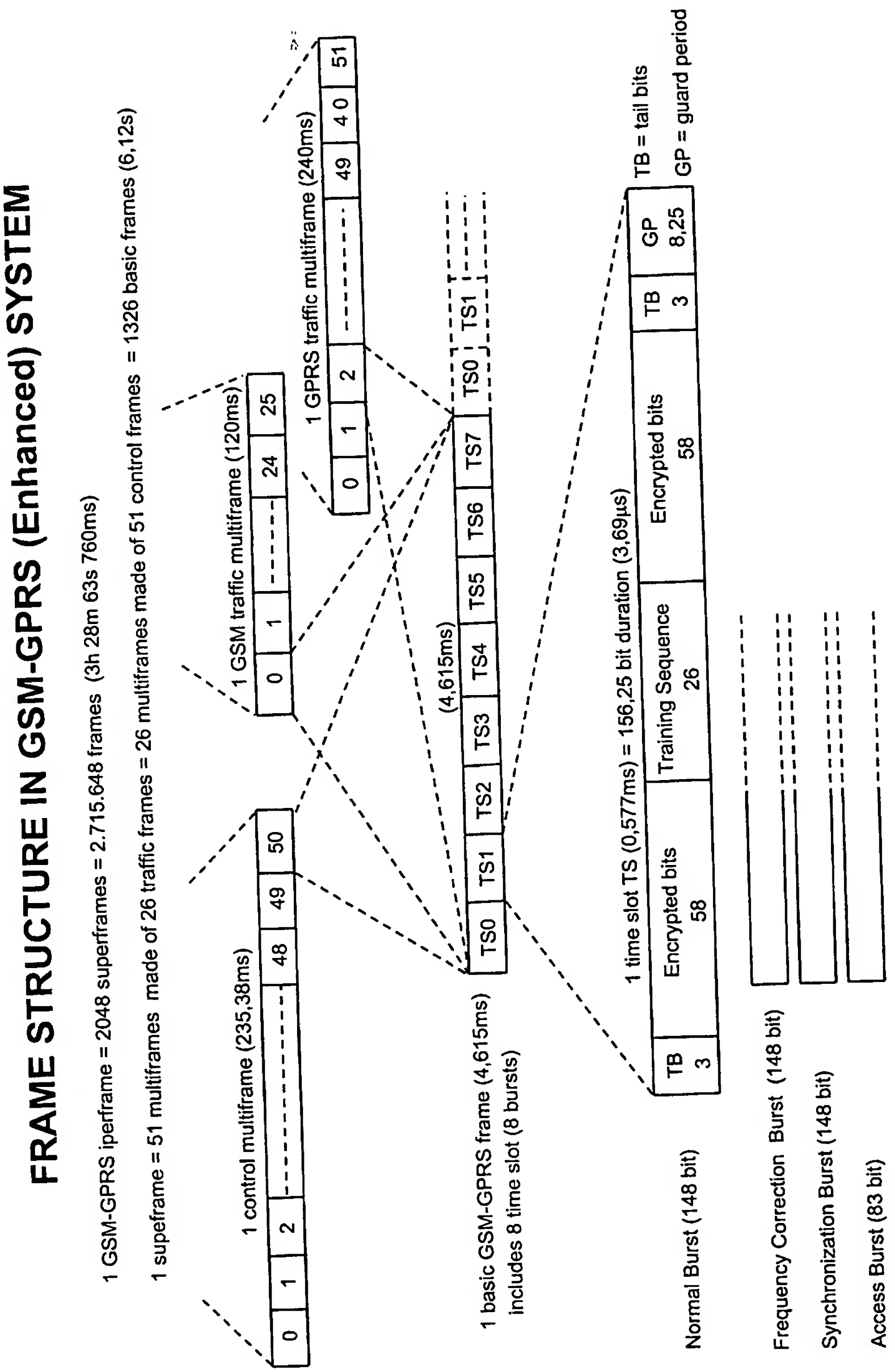

FRAME STRUCTURE IN GSM-GPRS (Enhanced) SYSTEM
1 GSM-GPRS iperframe = 2048 superframes = 2.715.648 frames (3h 28m 63s 760ms)
1 supeframe = 51 multiframes made of 26 traffic frames = 26 multiframes made of 51 control frames = 1326 basic frames (6,12s)
1 control multiframe (235,38ms)
0
1
2
48
49
50
1 GSM traffic multiframe (120ms)
0
1
24
25
1 GPRS traffic multiframe (240ms)
0
1
2
49
40
51
1 basic GSM-GPRS frame (4,615ms) includes 8 time slot (8 bursts)
(4,615ms)
TS0
TS1
TS2
TS3
TS4
TS5
TS6
TS7
TS0
TS1
1 time slot TS (0,577ms) = 156,25 bit duration (3,69µs)
Normal Burst (148 bit)
TB
3
Encrypted bits
58
Training Sequence
26
Encrypted bits
58
TB
3
GP
8,25
TB = tail bits
GP = guard period
Frequency Correction Burst (148 bit)
Synchronization Burst (148 bit)
Access Burst (83 bit)

FIG.2

TRAFFIC CHANNEL ORGANIZATION

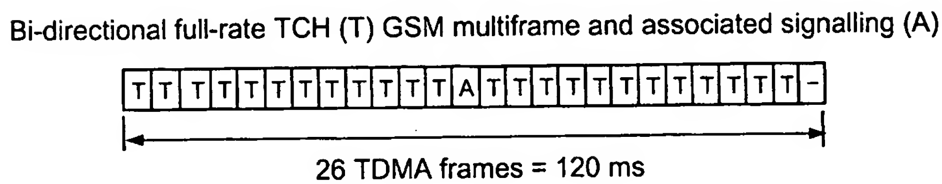


FIG.3a

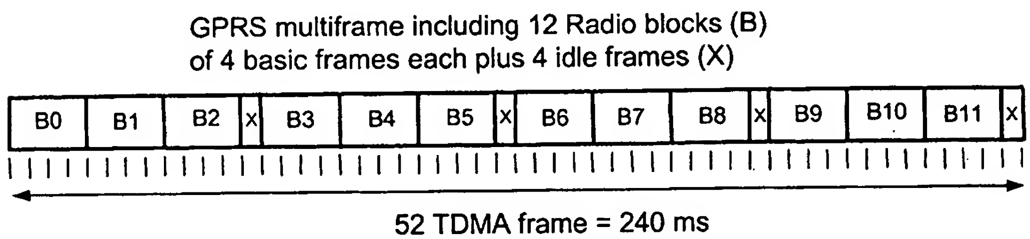


FIG.3b

MAPPING RLC LAYER INTO PHYSICAL LAYER

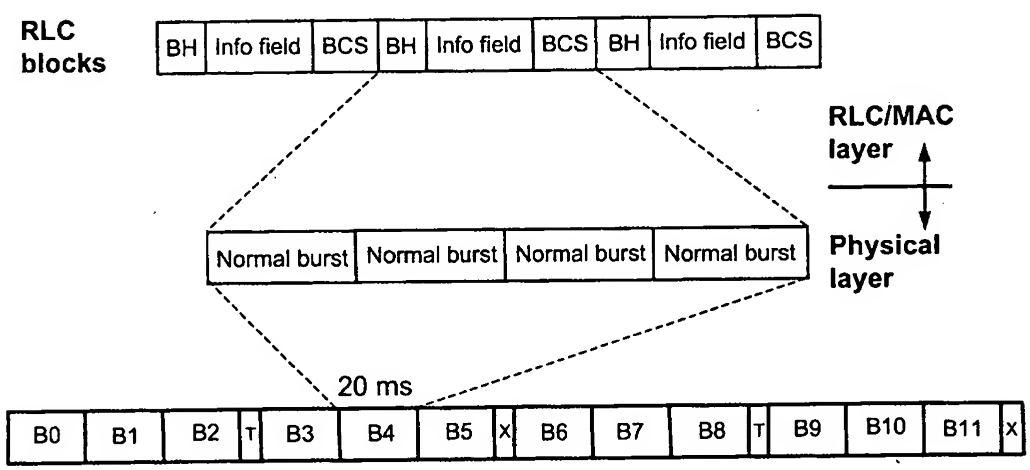


FIG.4

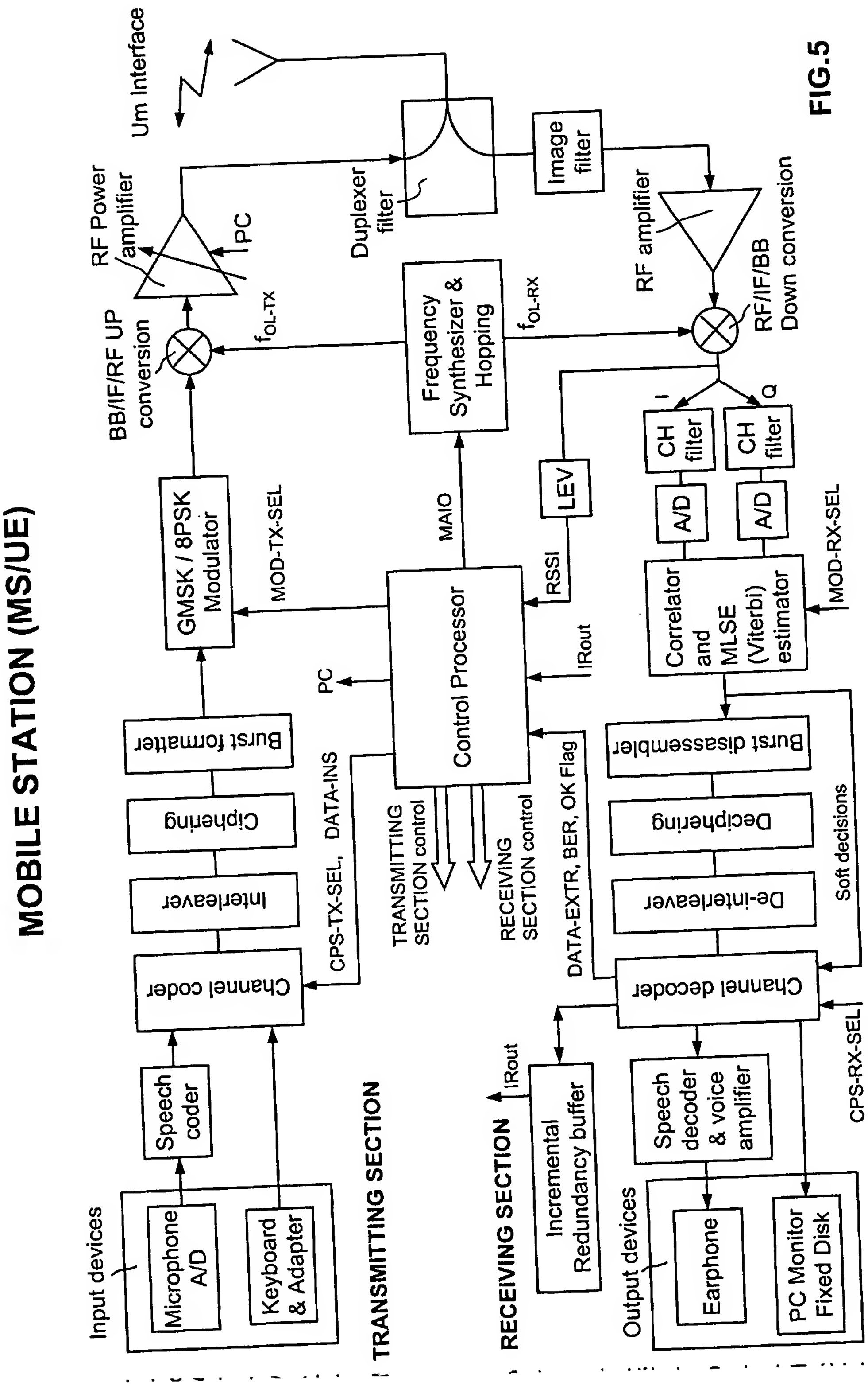

MOBILE STATION (MS/UE)
Um Interface
RF Power amplifier
PC
Duplexer filter
Image filter
BB/IF/RF UP conversion
f_{OL-TX}
Frequency Synthesizer & Hopping
f_{OL-RX}
RF amplifier
RF/IF/BB Down conversion
I
Q
CH filter
CH filter
A/D
A/D
LEV
GMSK / 8PSK Modulator
MOD-TX-SEL
MAIO
RSSI
Control Processor
Correlator and MLSE (Viterbi) estimator
MOD-RX-SEL
IRout
PC
Burst formatter
Ciphering
Interleaver
Channel coder
CPS-TX-SEL, DATA-INS
TRANSMITTING SECTION control
RECEIVING SECTION control
DATA-EXTR, BER, OK Flag
Burst disassembler
Deciphering
De-interleaver
Channel decoder
Soft decisions
CPS-RX-SEL
Speech coder
Input devices
Microphone A/D
Keyboard & Adapter
TRANSMITTING SECTION
RECEIVING SECTION
IRout
Incremental Redundancy buffer
Speech decoder & voice amplifier
Output devices
Earphone
PC Monitor Fixed Disk

FIG.5

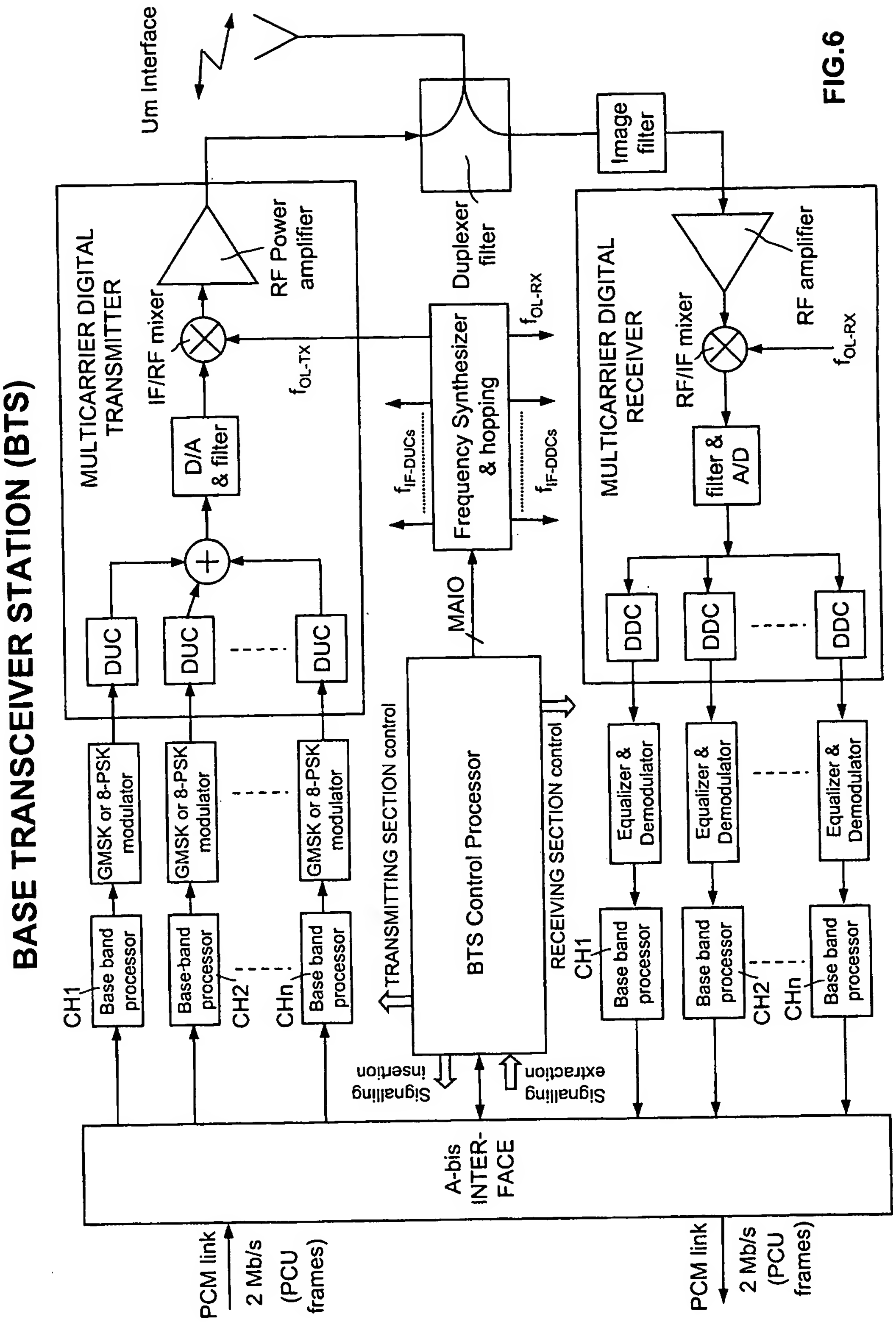

BASE TRANSCEIVER STATION (BTS)
Um Interface
MULTICARRIER DIGITAL TRANSMITTER
IF/RF mixer
RF Power amplifier
D/A & filter
DUC
DUC
DUC
f_{OL-TX}
Duplexer filter
Image filter
Frequency Synthesizer & hopping
$f_{IF-DUCs}$
$f_{IF-DDCs}$
f_{OL-RX}
MAIO
MULTICARRIER DIGITAL RECEIVER
RF/IF mixer
RF amplifier
f_{OL-RX}
filter & A/D
DDC
DDC
DDC
GMSK or 8-PSK modulator
GMSK or 8-PSK modulator
GMSK or 8-PSK modulator
CH1
Base band processor
Base-band processor
CH2
CHn
Base band processor
TRANSMITTING SECTION control
BTS Control Processor
RECEIVING SECTION control
Equalizer & Demodulator
Equalizer & Demodulator
Equalizer & Demodulator
CH1
Base band processor
Base band processor
CH2
CHn
Base band processor
Signalling insertion
Signalling extraction
A-bis INTER-FACE
PCM link
2 Mb/s (PCU frames)
PCM link
2 Mb/s (PCU frames)

FIG.6

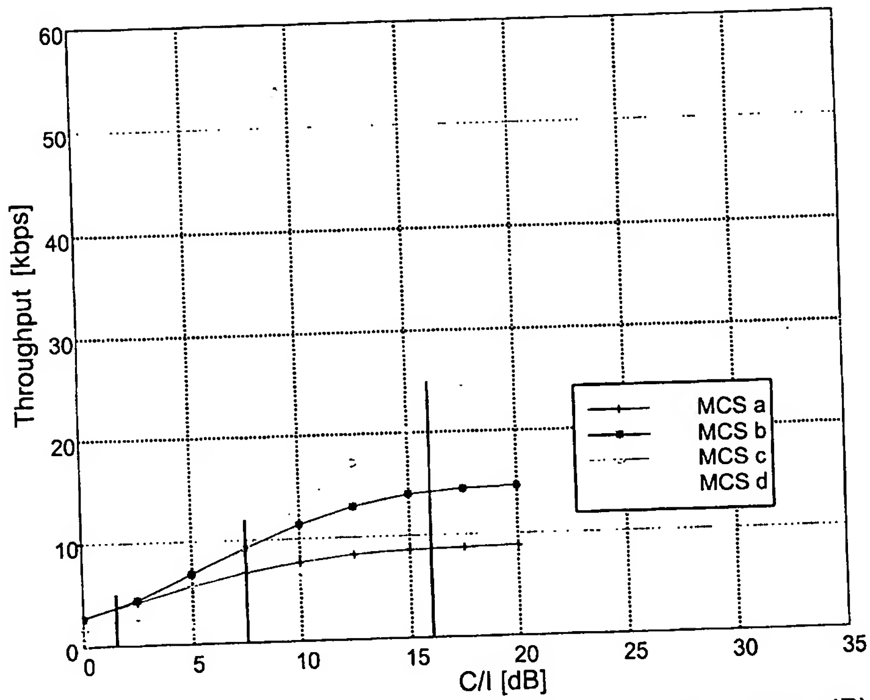


FIG.7 Simulation results for a selection of MCS (low diversity, without IR)

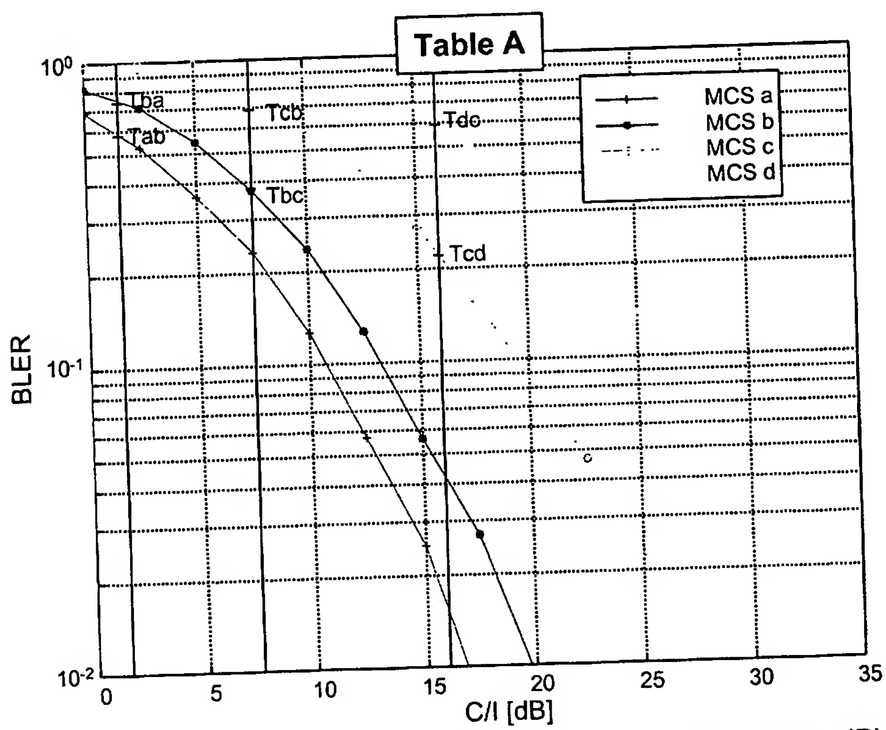


FIG.8 BLER versus C/I for a selection of MCS (low diversity, without IR)

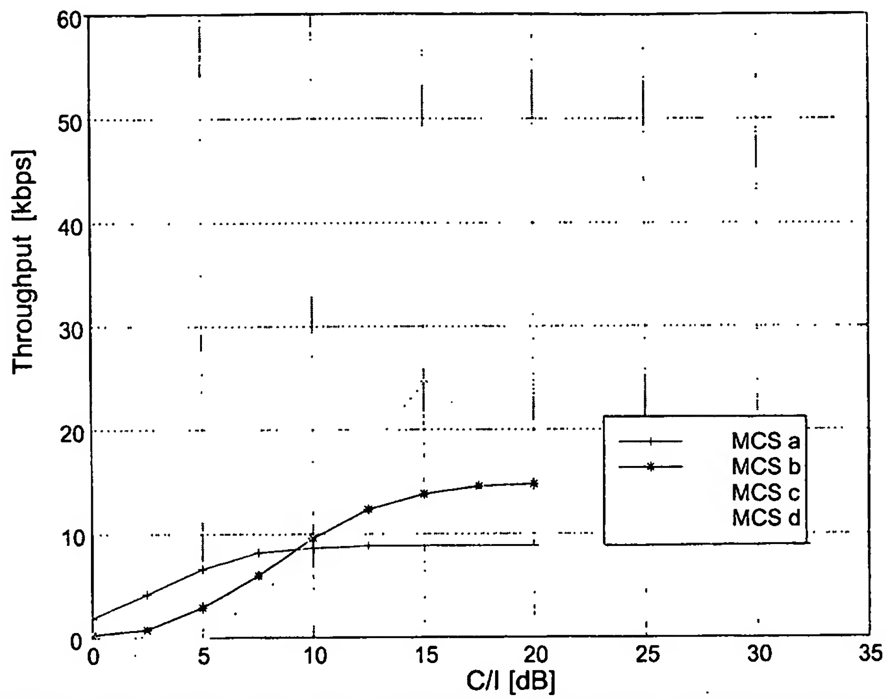


FIG.9 Simulation results for a selection of MCS (high diversity, without IR)

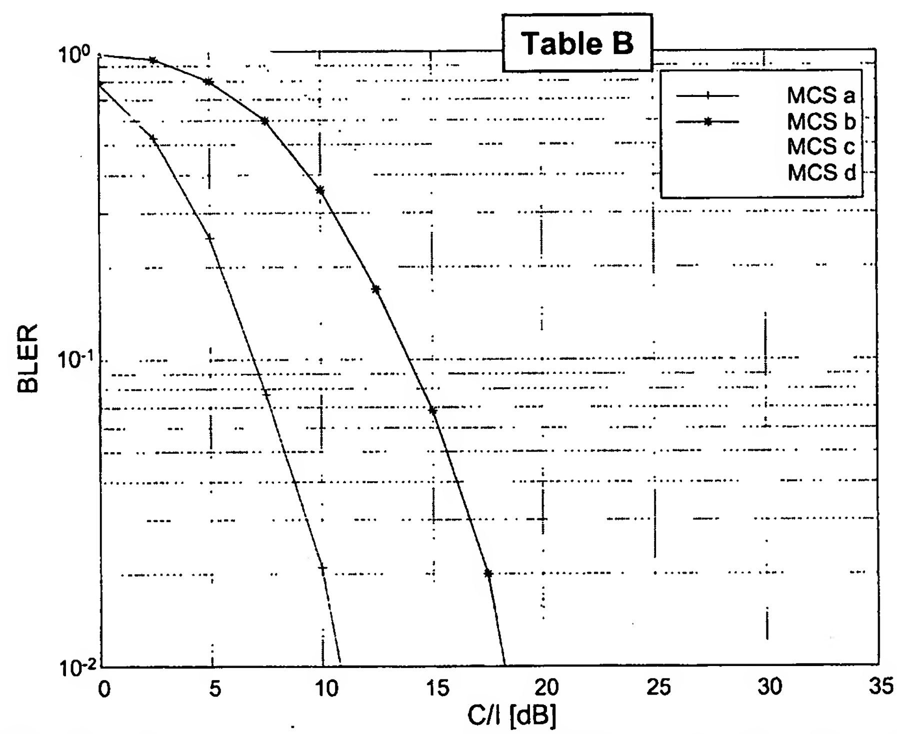


FIG.10 BLER versus C/I for a selection of MCS (high diversity, without IR)

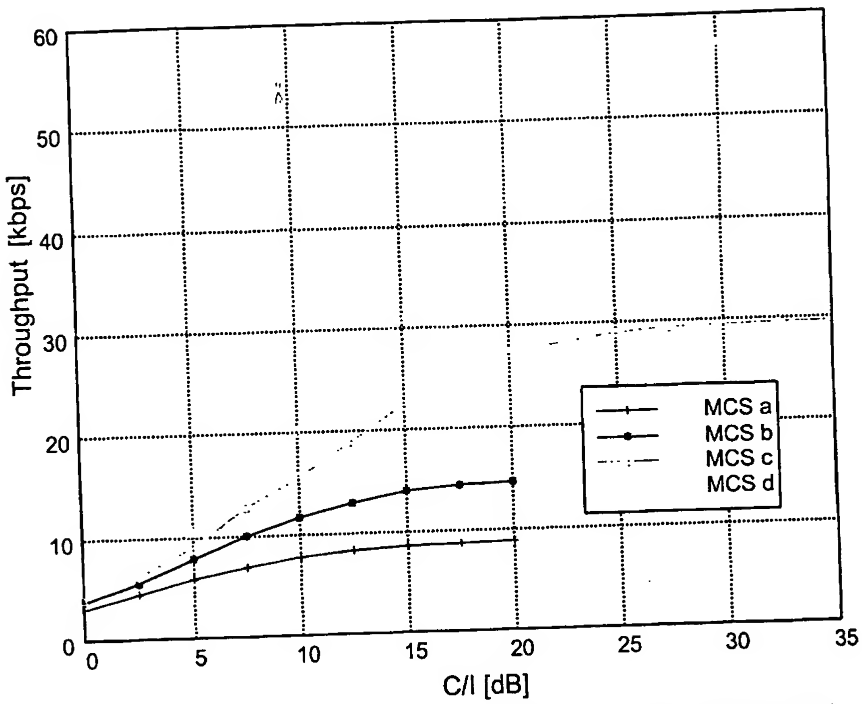


FIG.11 Simulation results for a selection of MCS (low diversity, with IR)

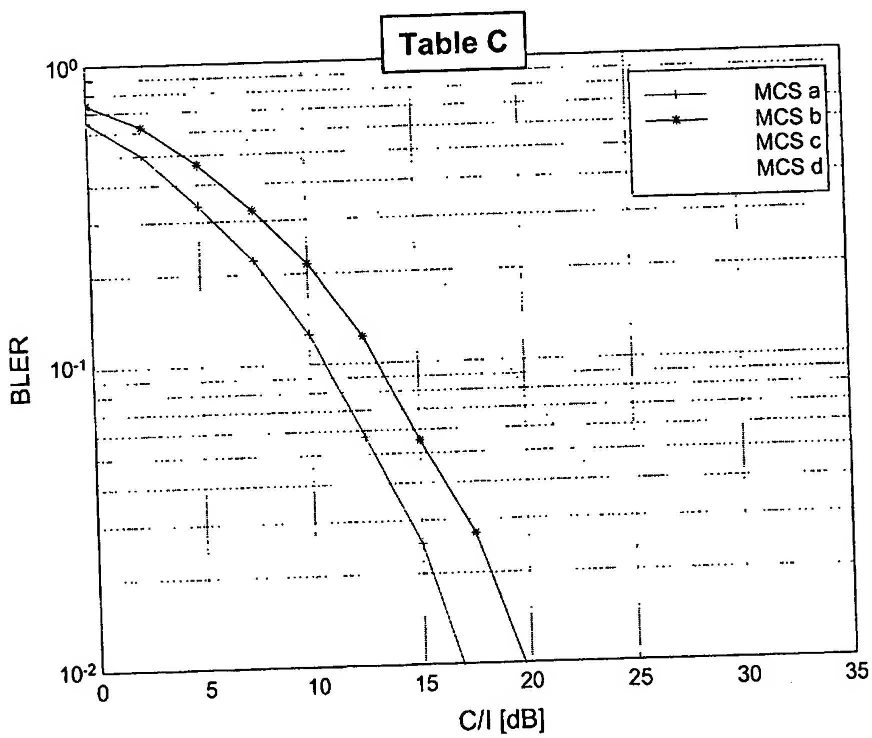


FIG.12 BLER versus C/I for a selection of MCS (low diversity, with IR)

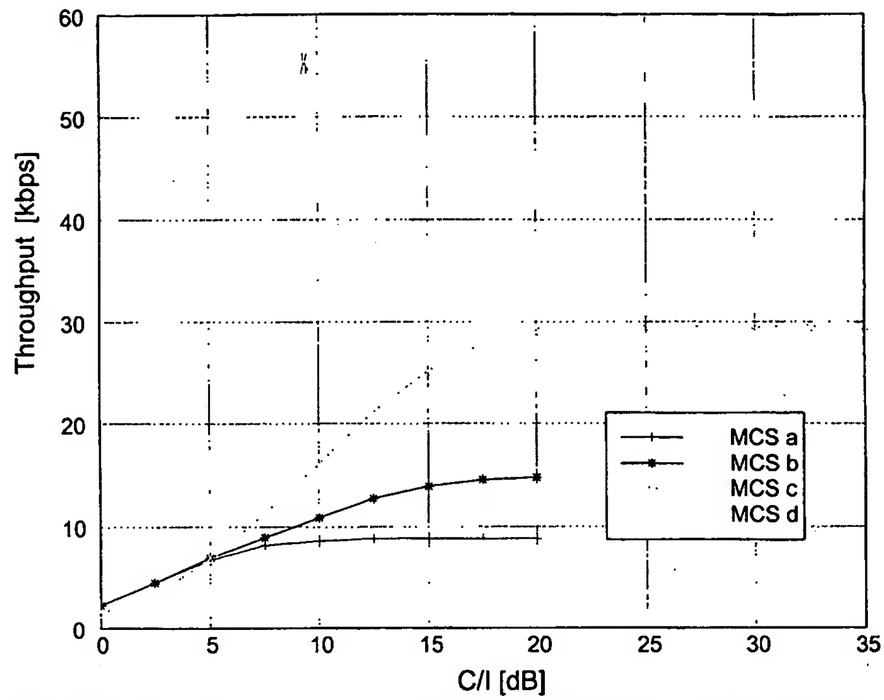


FIG.13 Simulation results for a selection of MCS (high diversity, with IR)

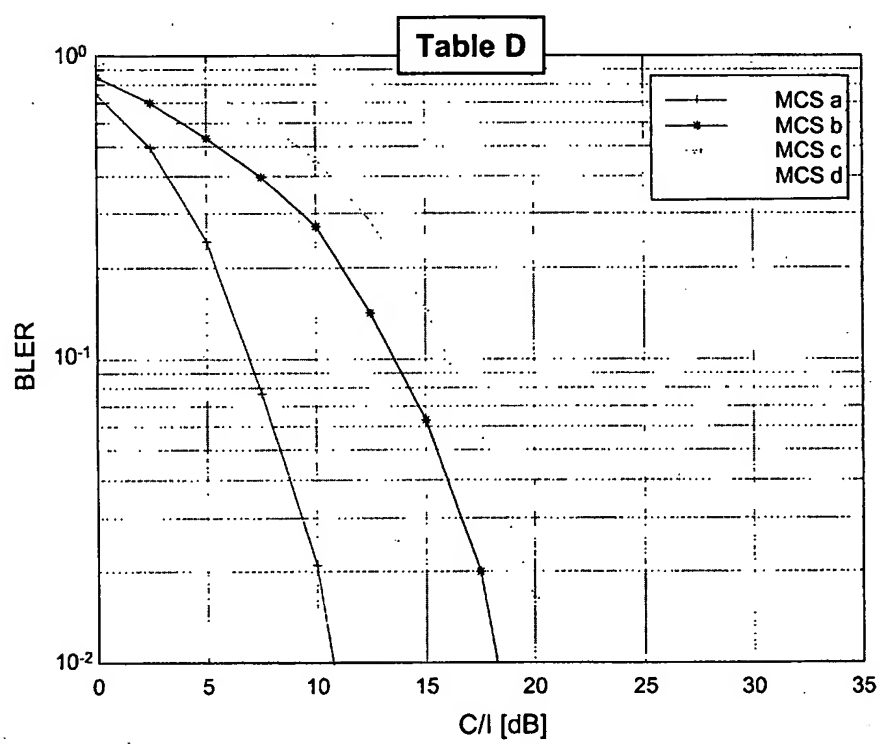


FIG.14 BLER versus C/I for a selection of MCS (high diversity, with IR)

METHOD TO PERFORM LINK ADAPTATION WITHOUT IR

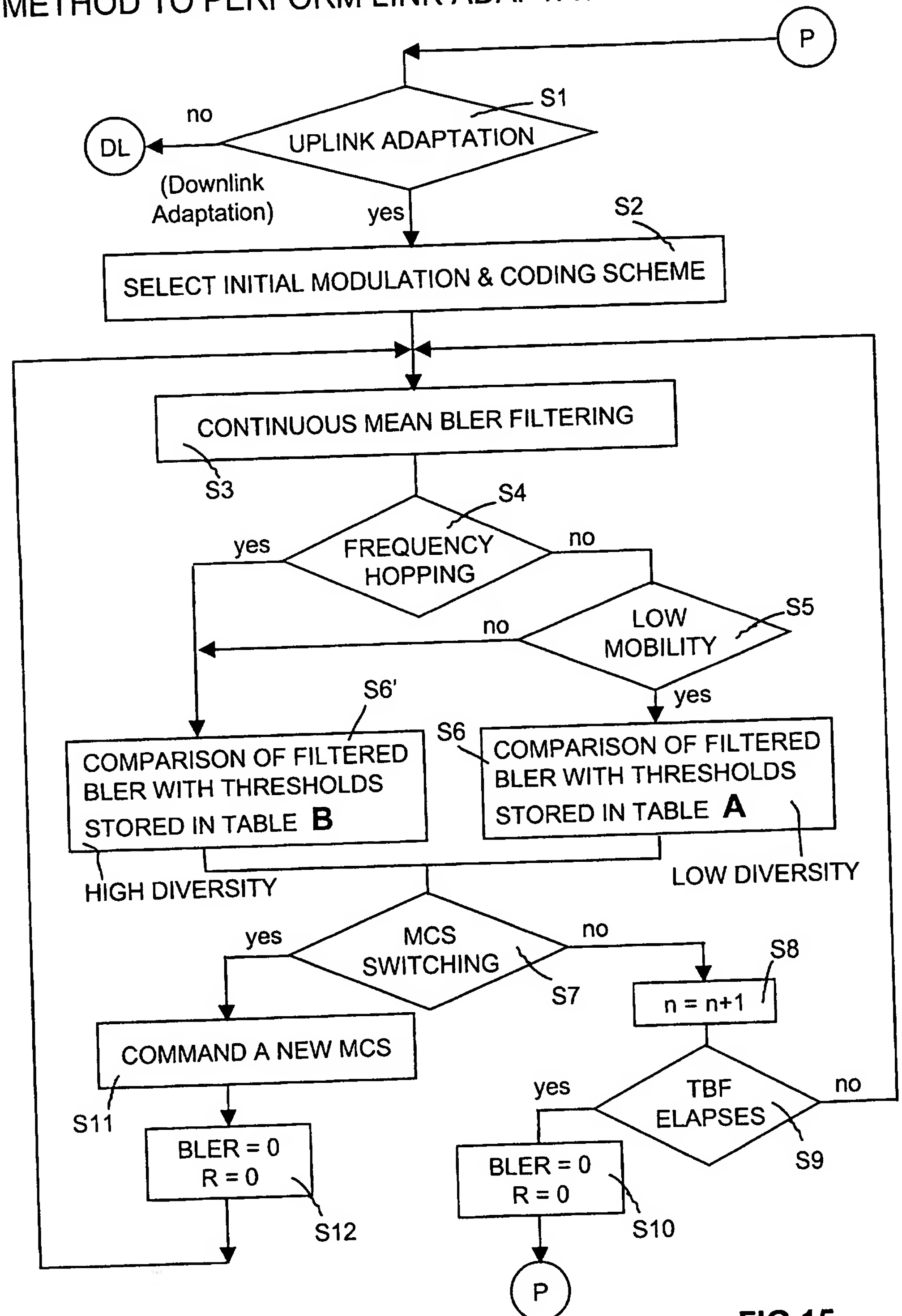


FIG.15

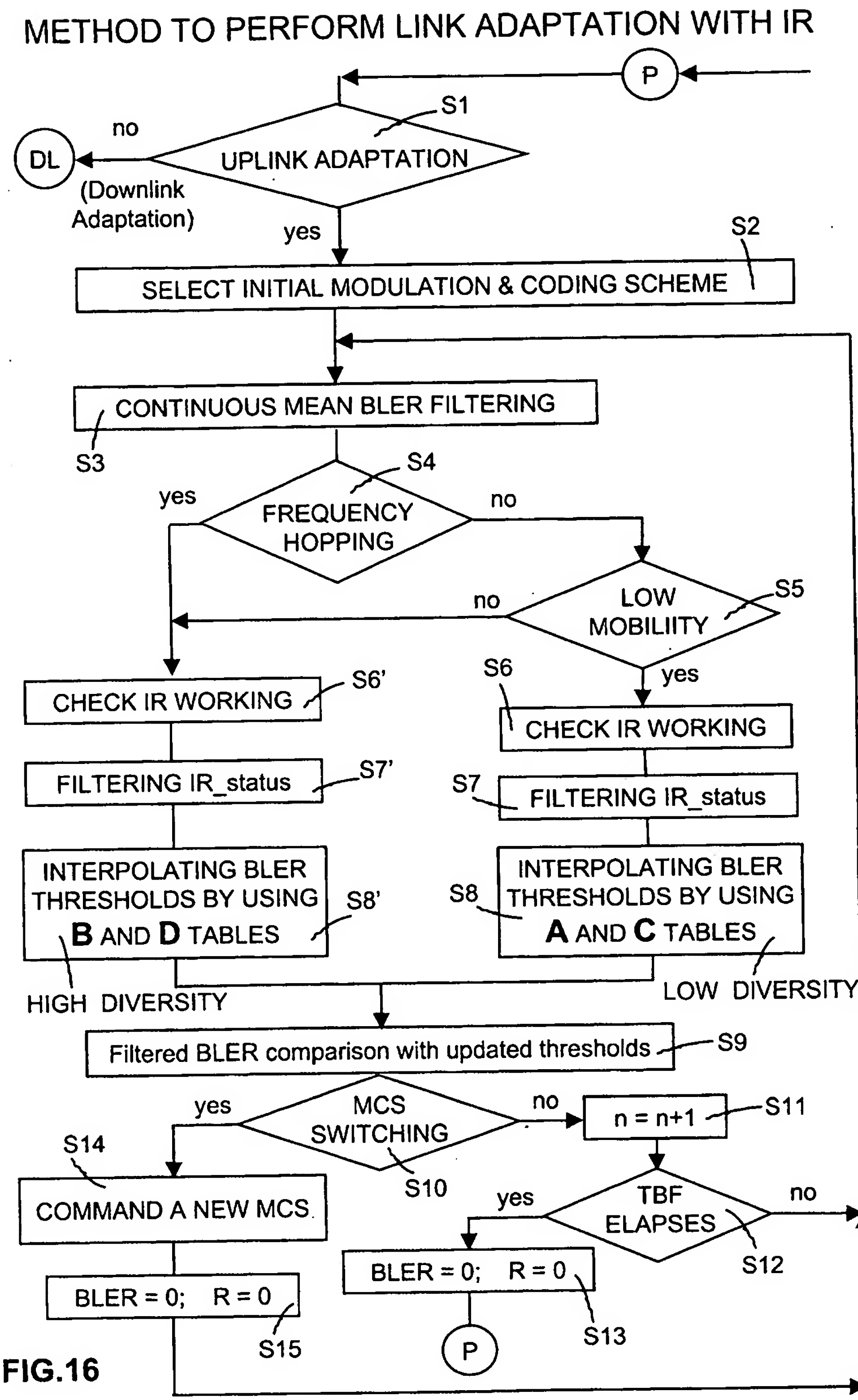
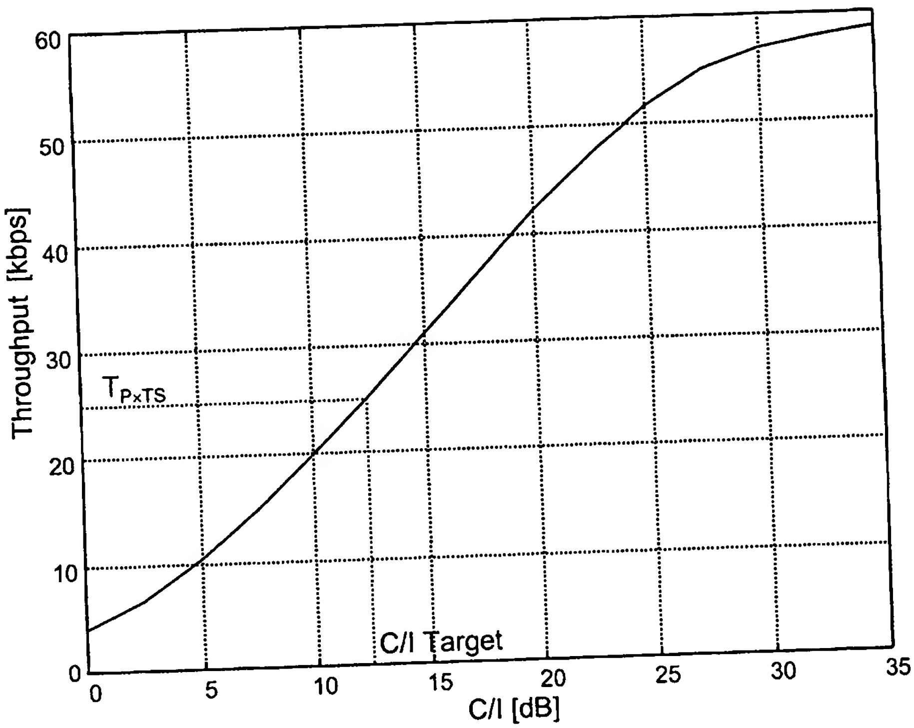

METHOD TO PERFORM LINK ADAPTATION WITH IR
P
S1
UPLINK ADAPTATION
no
DL
(Downlink Adaptation)
yes
S2
SELECT INITIAL MODULATION & CODING SCHEME
CONTINUOUS MEAN BLER FILTERING
S3
S4
FREQUENCY HOPPING
yes
no
LOW MOBILIITY
S5
no
yes
CHECK IR WORKING
S6'
S6
CHECK IR WORKING
FILTERING IR_status
S7'
S7
FILTERING IR_status
INTERPOLATING BLER THRESHOLDS BY USING B AND D TABLES
S8'
S8
INTERPOLATING BLER THRESHOLDS BY USING A AND C TABLES
HIGH DIVERSITY
LOW DIVERSITY
Filtered BLER comparison with updated thresholds
S9
MCS SWITCHING
yes
no
n = n+1
S11
S14
S10
COMMAND A NEW MCS
TBF ELAPSES
yes
no
S12
BLER = 0; R = 0
S13
P
BLER = 0; R = 0
S15

FIG.16



Maximum achievable throughput (with IR)

FIG.17

European Patent Office

EUROPEAN SEARCH REPORT

Application Number
EP 01 83 0283

DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (Int.Cl.7)
X	US 5 764 699 A (CRISLER KENNETH J ET AL) 9 June 1998 (1998-06-09)	1	H04L1/00
Y	* abstract * * figures 3,4 * * column 5, line 59 - column 7, line 27 * ---	2-4,7,9	
Y	SHELDON M. ROSS: "Introduction to Probability and Statistics for Engineers and Scientists" 1987 , JOHN WILEY&SONS , NEW YORK XP002180737 * page 426 - page 427 * ---	2-4,7,9	
A,D	WO 99 12304 A (ERICSSON TELEFON AB L M) 11 March 1999 (1999-03-11) * the whole document * -----	1-10	

TECHNICAL FIELDS SEARCHED (Int.Cl.7)
H04L

~~The present search report has been drawn up for all claims~~

Place of search	Date of completion of the search	Examiner
THE HAGUE	29 January 2002	Borges, P

CATEGORY OF CITED DOCUMENTS

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EPO FORM 1503 03.82 (P04C01)

ANNEX TO THE EUROPEAN SEARCH REPORT ON EUROPEAN PATENT APPLICATION NO.

EP 01 83 0283

This annex lists the patent family members relating to the patent documents cited in the above-mentioned European search report.
The members are as contained in the European Patent Office EDP file on
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EPO FORM P0459

For more details about this annex : see Official Journal of the European Patent Office, No. 12/82

♦ A Proposal for EGPRS Radio Link Control Using Link Adaptation and Incremental Redundancy

Krishna Balachandran, Keith F. Conner, Richard P. Ejzak, and Sanjiv Nanda

Standardization is currently in progress to specify Enhanced Data rates for GSM Evolution (EDGE). The two components of EDGE, Enhanced Circuit-Switched Data (ECSD) and Enhanced General Packet Radio Service (EGPRS), define enhancements for circuit-mode and packet-mode data, respectively. EGPRS is also intended to provide a high-rate North American digital time division multiple access (TDMA) packet data service. To achieve higher data rates, 8-ary phase shift keying (8-PSK) modulation has been introduced. Enhanced radio link control (RLC) procedures are being defined based on link adaptation and incremental redundancy to achieve superior delay/throughput performance over a wide range of operating conditions. This paper describes a candidate proposal for (RLC) block segmentation for EGPRS that provides combined link adaptation and incremental redundancy; short, fixed-length blocks to achieve lower block error rates; seamless transition between coding schemes without loss in throughput; segmentation to provide a robust Gaussian minimum shift keying (GMSK) fallback mode; and short and extended header formats. We present performance tradeoffs for the proposed scheme. Many of the features of this proposal have been included in the EGPRS specification. We provide a brief description of the block and header formats being standardized.

Introduction

Enhanced General Packet Radio Service (EGPRS) is currently being standardized for the evolution of current Global System for Mobile Communications (GSM) systems to support high-rate packet data services. EGPRS can support higher data rates compared to basic General Packet Radio Service (GPRS) through the use of 8-ary phase shift keying (8-PSK) modulation in addition to Gaussian minimum shift keying (GMSK) modulation.

Radio link and medium access protocols based on incremental redundancy and dynamic link adaptation between different coding and modulation schemes are of great interest for EGPRS in order to achieve superior delay/throughput performance over a wide range of operating conditions. In the following sections, we propose a simple transmitter structure that enables both link adaptation and incremental redundancy for EGPRS.

Background: Incremental Redundancy in GPRS-136

Packet data services known as GPRS and GPRS-136 (formerly known as IS-136+) have been defined for GSM and IS-136 time division multiple access (TDMA) standards, respectively. GPRS-136[1] employs *adaptive modulation* and *incremental redundancy* to achieve higher throughput. For incremental redundancy, each GPRS-136 radio link protocol (RLP) data block is coded into D data sub-blocks and D parity sub-blocks. A rate 1/2 nonsystematic convolutional

"mother" code is used, then the output of one generator is mapped into D "data" sub-blocks and the output of the other generator is mapped into D "parity" sub-blocks. The D data sub-blocks are transmitted first, resulting in a code rate of unity for the first transmission. In response to each negative acknowledgment, one parity sub-block is transmitted to achieve the code rates: $D/(D+1)$, ..., 1/2. Thus incremental redundancy efficiently matches the effective code rate to the channel signal-to-interference-and-noise ratio (SINR). The parameter, D, may be considered as an incremental redundancy code rate *granularity* parameter. Incremental redundancy with a rate 1/2 mother code and granularity $D = 3$ has been specified in the GPRS-136 standard, while EGPRS will use a rate 1/3 mother code and incremental redundancy with granularity $D = 1$. With $D = 1$, if the first transmission is uncoded, then the first and second redundancy transmissions result in effective code rates of 1/2 and 1/3, respectively.

In the case of incremental redundancy, soft information corresponding to received sub-blocks must be combined with soft information obtained from previously received sub-blocks for successful decoding. Therefore, the receiver stores soft information corresponding to data that has not yet been decoded successfully. Furthermore, the headers containing sequence numbers must be separately coded in order to identify data/parity sub-blocks at the receiver. The receiver must first decode the header and determine how to soft-combine the received sub-block with previously stored information in order to decode the data segment successfully.

For adaptive modulation in GPRS-136, the modulation is switched between 4-, 8-, and 16-level modulation as a function of the SINR. Only $\pi/4$ differential quaternary PSK (DQPSK) and coherent 8-PSK have been standardized at this point, with the possibility of a 16-level modulation to be standardized in the future. These modulations allow the transmission of 2, 3, and 4 bits per symbol, respectively. To achieve this with a fixed number of symbols per frame, 2, 3, or 4 RLP data/parity sub-blocks are mapped to the TDMA slot. Associated with each slot (2, 3, or 4 data/parity sub-blocks) is a data segment header, which contains the 10-bit sub-block sequence number and a parity/control sub-block pointer (PCBP) to indicate the composition of the slot in terms of data and parity/control sub-blocks (2 bits). A 12-bit cyclic redundancy check (CRC) sequence is added, and the resulting header is encoded using a punctured, tailbiting, nonsystematic, 32-state, rate 1/2 convolutional code to obtain the coded data segment header.

Panel 1. Abbreviations, Acronyms, and Terms

AWGN—additive white Gaussian noise
BLER—block error rate
C/N—carrier-to-noise ratio
CR—code rate
CRC—cyclic redundancy check
CSN—coded sub-block number
DQPSK—differential quaternary phase shift keying
ECSD—Enhanced Circuit-Switched Data
EDGE—Enhanced Data rates for GSM Evolution
EGPRS—Enhanced GPRS
ETSI—European Telecommunications Standards Institute
GMSK—Gaussian minimum shift keying
GPRS—General Packet Radio Service
GPRS-136—General Packet Radio Service-136 (packet data standard for TDMA-based wireless systems)
GSM—Global System for Mobile Communications
IR—incremental redundancy
IS-136—Interim Standard-136 (U.S. standard for TDMA-based wireless systems)
LA—link adaptation
LLC—logical link control
MAC—media access control
MCS—modulation and coding scheme
PCBP—parity/control sub-block pointer
PSK—phase shift keying
RF—radio frequency
RLC—radio link control
RLP—radio link protocol
SACCH—slow associated control channel
SB—stealing bit
SINR—signal-to-interference-and-noise ratio
TDMA—time division multiple access
TU—transmission unit
TU3—typical urban channel model as specified in GSM with a mobile speed of 3 km/h
USF—uplink state flag

GPRS-136 also specifies a more conventional "fixed" coding mode. Here, separately coded headers are not required. The headers and data are encoded together using the same convolutional code. Further details of GPRS-136 are described by Balachandran et al.[2]

Proposal: Incremental Redundancy for EGPRS

The EGPRS radio link control (RLC)/media access control (MAC) layer is required to provide reliable in-sequence delivery of data provided to it by the logical link control (LLC) layer (that is, layer 3 of the GPRS/EGPRS protocol stack). For EGPRS RLC standardization, several segmentation schemes that enable incremental redundancy and link adaptation were proposed.[3,4,5] The basic unit of transmission in EGPRS is a 20-ms EGPRS *radio block* spanning 4 GSM physical layer bursts on a given time slot. Investigations of the performance of these schemes showed that the use of a *short*, fixed-length RLC block size, where multiple RLC blocks are mapped to an EGPRS radio block, results in higher mean throughput compared to the use of a single RLC block of variable size per EGPRS radio block. However, the peak throughput is somewhat reduced with short blocks because of the increase in header and CRC overhead.

This paper discusses in detail the RLC block segmentation procedure Lucent Technologies proposed to the European Telecommunications Standards Institute (ETSI).[4] The proposed scheme is shown to provide the best throughput over a wide range of operating conditions. The segmentation scheme permits higher granularity of coding rates with incremental redundancy ($D = 2$) to achieve performance that is well matched to the prevailing SINR. Performance tradeoffs for block size and coding rate are presented. The features of the scheme are as follows:

- Combined link adaptation (LA) and incremental redundancy (IR);
- *Short*, fixed-length blocks to achieve lower block error rates (BLERs);
- Mapping of an RLC block to one GSM physical layer burst in the uncoded 8-PSK case (this provides lower BLER with frequency hopping via single slot interleaving as described in the Nokia proposal[5]);
- Seamless transition between coding schemes without loss in throughput;
- Resegmentation to provide a robust GMSK fallback mode as in the Ericsson proposal;[3] and
- Short and extended header formats that are reliably coded by decreasing the amount of coding or the redundant information provided on retransmissions.

A joint proposal for RLC segmentation and header formats was developed by several companies (AT&T, Ericsson, Lucent, Nokia, and Nortel) and became the basis for the RLC specification for EGPRS. The details of the RLC segmentation scheme and header formats for the standard are described at the end of this paper. The standard will incorporate several simplified features described in this paper (and in Lucent's proposal to ETSI[4]). For example:

- The standard specifies several families of RLC block sizes rather than the one fixed block size proposed here. This was determined to be an effective compromise between the BLER advantage of the smaller blocks and the overhead advantage of fewer blocks per EGPRS radio block.
- The RLC block sizes and segmentation schemes as specified in the standard are restricted to $D = 1$. Larger values of D (for example, $D = 3$ in GPRS-136 and $D = 2$ in the proposal described in this paper) allow more code rate granularity for IR.
- The standard specifies the use of multiple header formats for use with different modulation and coding schemes, as proposed in this paper.
- When longer headers are needed, our proposal permits the dropping of coded parity subblocks in the IR mode. This permits more flexible adaptation (not restricted to families of formats) and $D = 2$. This option is not included in the standard.

The EGPRS RLC standards will be completed near the end of 1999.

Proposed Segmentation Scheme

The proposed segmentation scheme is shown in **Figures 1** and **2**. The LLC layer data is segmented into

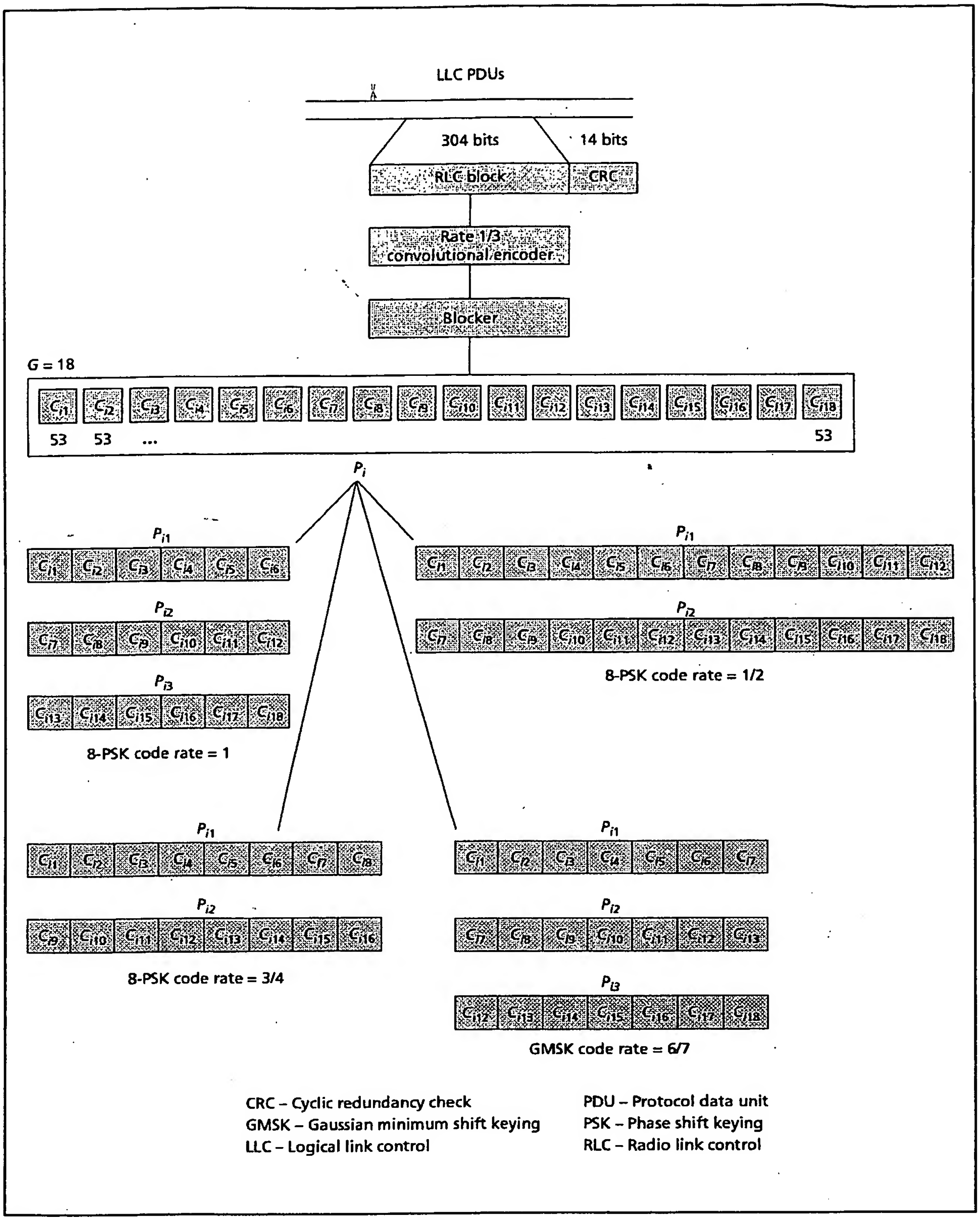


Figure 1.
Segmentation of RLC blocks for link adaptation and incremental redundancy D = 1, G = 18.

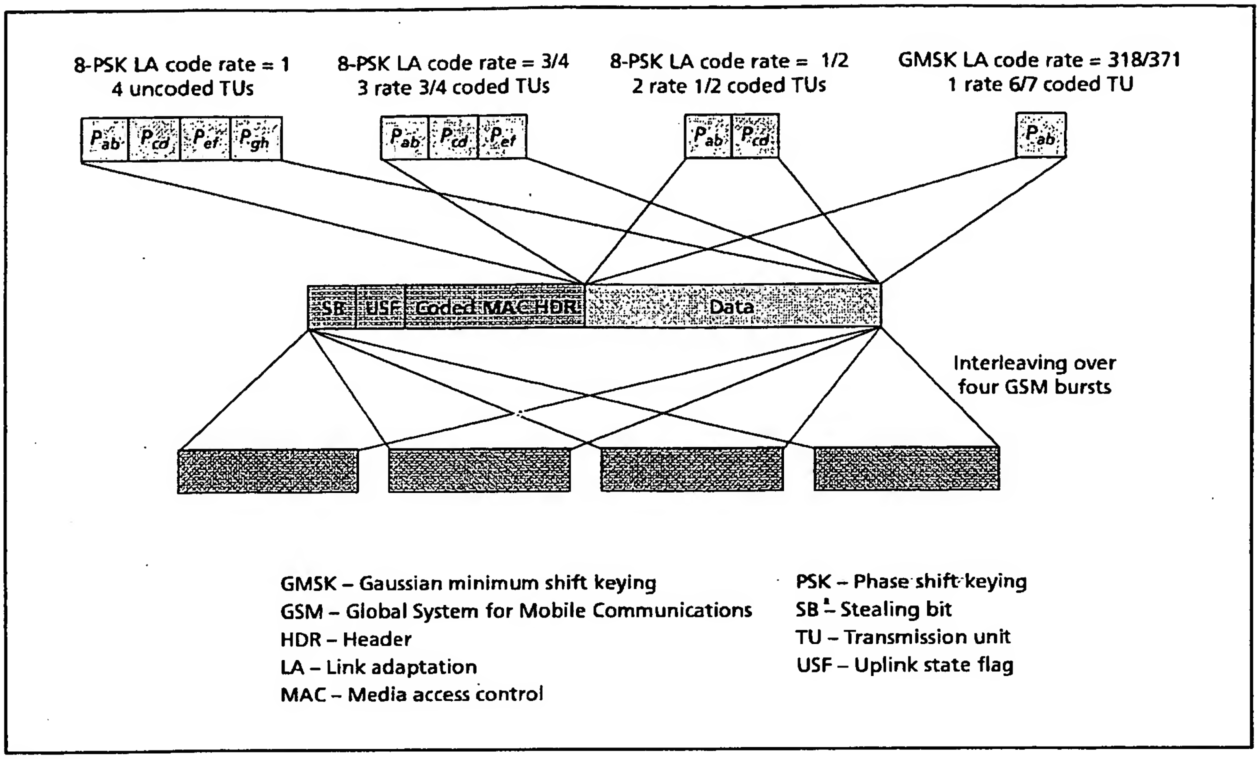


Figure 2.
Mapping of LA transmission units into 8-PSK downlink GSM bursts.

fixed-length *RLC blocks*, and a 14-bit CRC sequence is added to each block.

Each RLC block, B_i, is encoded using a rate 1/3 convolutional code. The encoder output is segmented into G coded sub-blocks, denoted C_{ij}, $j = 1, 2, ..., G$. A 12-bit coded sub-block number (CSN) is associated with each coded sub-block. In the following discussion, we have chosen $G = 18$, which enables code rates of 1, 3/4, and 1/2 with 8-PSK and 318/368 with GMSK. The segmentation into 18 sub-blocks allows the stealing of sub-blocks for the coding of extended header formats. The stealing of sub-blocks results in some increase in the LA code rate of the transmission unit (TU) from which the sub-block is dropped; with IR, only the amount of parity information available for soft combining is reduced.

Groups of coded sub-blocks are assembled to form TUs, P_{ik}, as shown in Figure 1. Depending on the current code rate of the LA scheme, each 8-PSK TU consists of 6, 8, or 12 consecutive coded sub-blocks associated with the same RLC block, which correspond to code rates of 1, 3/4, and 1/2, respectively. The TUs are of variable size. Thus, for $G = 18$, for code rates 1, 3/4, and 1/2, an 8-PSK TU consists of 6, 8, and 12 consecutive coded sub-blocks, respectively. A GMSK TU may be formed with 7 consecutive coded sub-blocks, resulting in a code rate of approximately 0.857.

Thus, the MAC header must explicitly or implicitly identify the CSN for up to 4 TUs. When the remaining coded sub-blocks in the TU are consecutive, the CSN of the first TU can be used to implicitly identify the remaining CSNs. The MAC header in this case explicitly identifies the CSN for the first of the TUs with consecutive CSNs. Short and extended header formats are defined later in the paper to accommodate a variable number of CSNs with an appropriate header coding.

On the downlink, multiple TUs are combined with other separately coded fields:

- A coded (short or extended) MAC header,
- An *uplink state flag* (USF) used to arbitrate reserved uplink access among active mobiles and designate contention slots, and

- *Stealing bits* (SBs) decoded by the receiver first to determine the composition of the headers and data.

The resulting segment is interleaved to obtain an EGPRS radio block that is transmitted over 4 GSM bursts, as shown in Figure 2. No interleaving is carried out in the uncoded 8-PSK case; for all other cases, GSM slow associated control channel (SACCH) interleaving is performed.

In all cases except uncoded 8-PSK, 4-slot interleaving is carried out as specified for the GSM SACCH.[6] In the case of uncoded 8-PSK, no interleaving of the data is performed and each TU is mapped into one GSM burst. The header is interleaved across 4 bursts.

Overhead and Peak Throughput Computation

The following sections describe overhead computations for 8-PSK and GMSK coding schemes.

8-PSK Coding Schemes

The RLC block sizes and peak throughput may be computed as follows:

- 8-PSK EGPRS radio block size (that is, number of bits per 4 GSM bursts) = 1,392 bits.
- USF = 36 bits.
- SB = 8 bits.
- Coded short MAC header size = 76 bits.

RLC block size = (1,392 – 36 – 8 – 76)/4
= 1,272/4 = 318 bits.

Number of bits per coded sub-block
= 318/6 = 53 bits.

Assuming a 14-bit CRC, we have 318 – 14
= 304 LLC data bits per RLC block.

Peak throughput = (304 × 4)/0.02 = 60.8 kb/s.

GMSK Coding Schemes

Two coding options are available with GMSK:

- The rate 318/371 (= 6/7) coding option allows the transmission of 7 coded sub-blocks per GMSK EGPRS radio block.
  - GMSK EGPRS radio block size = 464 bits.
  - USF = 12 bits.
  - SB = 8 bits.
  - Coded short MAC header size = 73 bits.

  Therefore, each RLC block has 318 bits, which are coded to 464 – 12 – 8 – 73 = 371 bits (318/371 = 6/7).
- The rate 195/444 (= 0.439) coding option (segmented RLC block) is shown in **Figure 3**.

Coded MAC Header Formats for 8-PSK

The following sections discuss header design principles and header formats for 8-PSK modulation.

Header Design Principles

Each TU included in the EGPRS radio block must be identified in the coded MAC header. Each TU is identified by the CSN of the first coded sub-block of the TU. When all TUs are consecutive, there is no need to include several CSNs in the coded MAC header. Therefore, when there are no retransmissions, the coded MAC header contains the CSN of the first TU only, and the remaining TUs are assumed to be consecutive. We refer to this header as the *short MAC header*. The short MAC header information bits are protected by an 8-bit CRC for header error detection. The 39-bit short MAC header (including CRC) is coded using a punctured rate 1/3 code to obtain 76 bits.

When there are retransmissions to be included, additional CSNs must be incorporated into the MAC header. This is done through the use of a series of extended MAC headers. Three types of extended MAC headers are defined. The lengths of the *coded* extended MAC header formats 1 and 2 are both 129 bits. Coded extended MAC header format 3 is of length 235 bits. The formats are designed to accommodate additional CSN fields for retransmissions as well as varying amounts of coding according to channel conditions. The length of coded extended MAC header formats is chosen to be 53 bits or 3 × 53 = 159 bits longer than the coded short MAC header (76 bits). The extra 53-bit multiples are obtained by dropping one or more coded sub-blocks from the TUs in the EGPRS radio block. (Each coded sub-block is 53 bits according to the RLC block size chosen).

In the case of 8-PSK, the size and the number of TUs are indicated implicitly through the use of the code rate (CR) field in the coded MAC header.

The extended header is only needed when there are retransmissions. For the IR case, the retransmitted TU can always be placed as the first TU in the EGPRS radio block. Dropping a coded sub-block reduces the amount of soft information in the first TU. Thus, for a

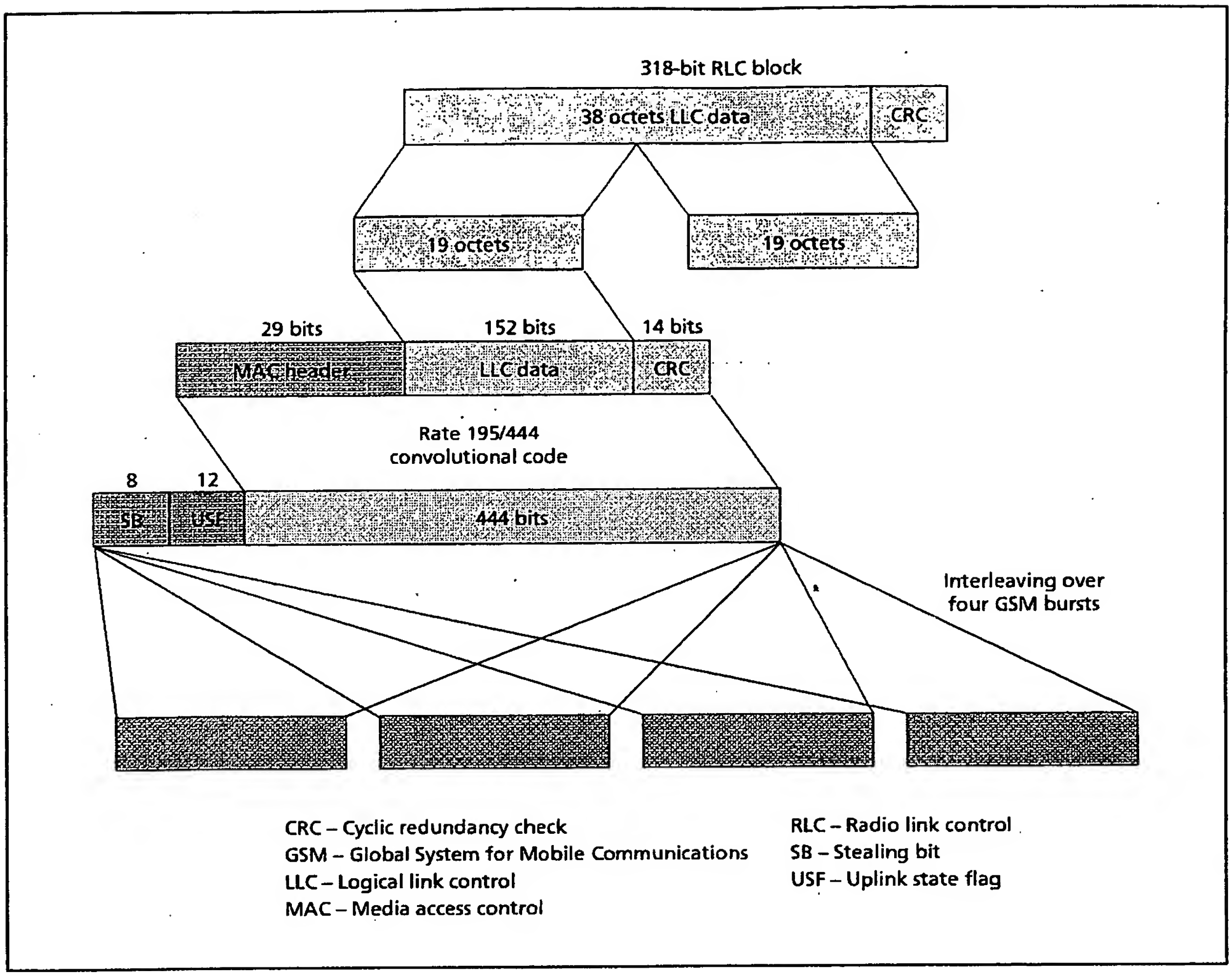


Figure 3.
Mapping of segmented RLC block into GMSK EGPRS radio block.

rate 3/4 (= 6/8) retransmission, dropping a coded sub-block makes the code rate of the transmitted TU to be 6/7. When the retransmitted TU is soft-combined with the earlier transmission in IR, the resultant rate becomes 6/15 (= 2/5) instead of 3/8 (= 6/16). Therefore, dropping sub-blocks is particularly well suited for IR. Similarly, when the first transmission is uncoded and one coded sub-block is dropped from the retransmission, we get a rate of 6/11 instead of 1/2 (= 6/12) on soft combining.

The coded extended MAC headers can be designed using the following principles:

1. The coding of the short MAC header at rate 1/2 is sufficient in good channel conditions, while in adverse channel conditions, the coded MAC header should be better protected. At the same time, the number of TUs per EGPRS radio block (and consequently the number of CSNs to be carried in the MAC header) is also smaller when using a lower code rate. The smaller number of CSNs permits better coding of the MAC header using the same number of coded bits.
2. When more than one CSN is included, then each corresponds to one TU in order except the last one, which signifies that the remaining TUs (with no CSN included in the coded MAC header) are in sequence. Thus, in extended MAC header format 1, the first CSN may correspond to a

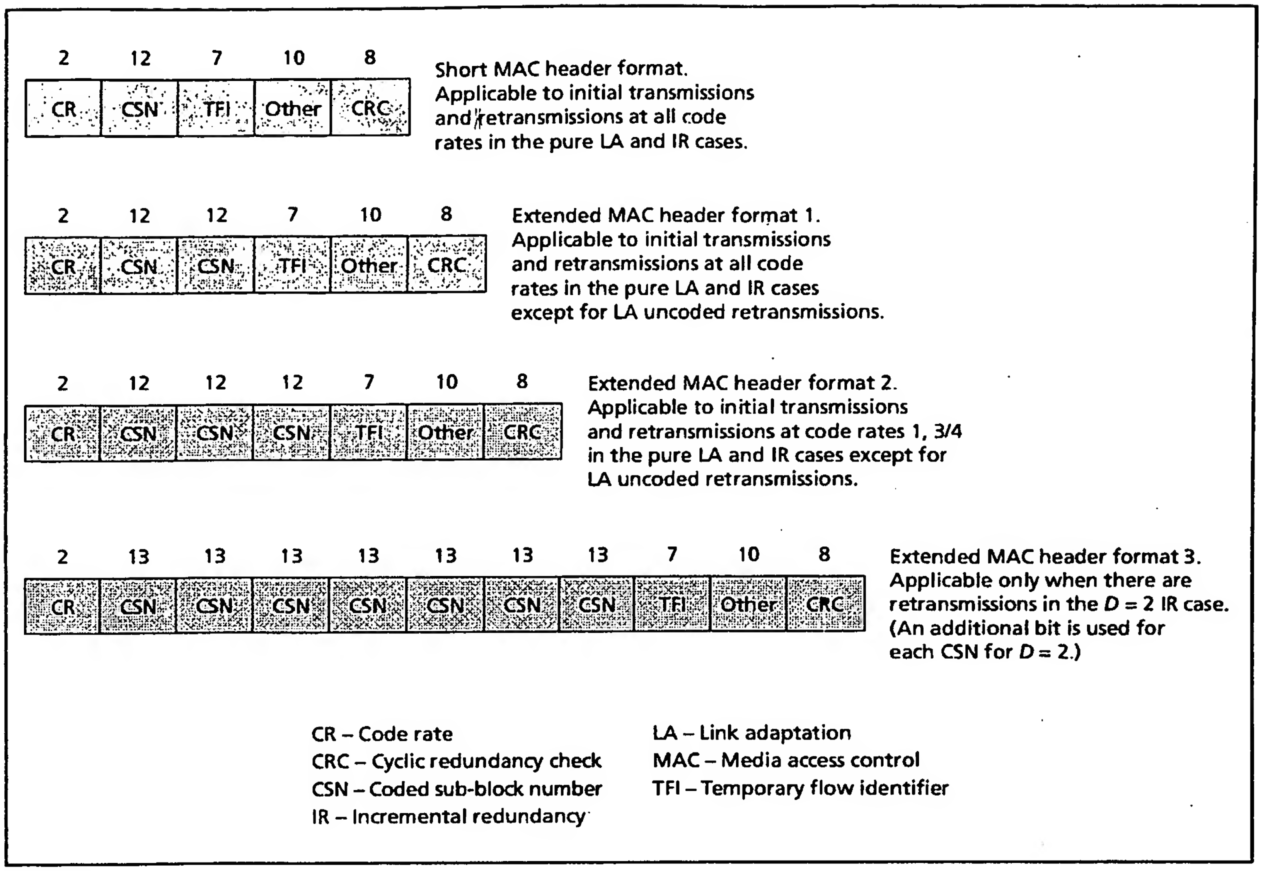


Figure 4.
MAC header formats for 8-PSK.

retransmitted TU, and the second one would indicate that remaining TUs are in sequence starting from the second CSN.

3. For IR, the retransmitted TUs are placed first and allow the transmission of less parity information on dropping coded sub-blocks for extended headers.

In particular, using the three principles above, we propose the simple formats shown in **Figure 4** as example MAC header formats. The coded MAC header format for each 8-PSK EGPRS radio block is indicated to the receiver through stealing bits. Within each format, the number and size of CSN fields are determined by the code rate (CR) field.

Header Formats

Short and extended header formats 1 and 2 are used for initial transmissions and retransmissions at the prevailing code rate in the pure LA case (that is, no soft combining at the receiver) and for initial transmissions and retransmissions in the IR case. In the pure LA case, retransmissions are at rate 3/4 using extended header formats 1 and 2 if the prevailing code rate is 1. Extended header format 3 is used only when there are retransmissions in the $D = 2$ IR case. In this case, the CR field does not bear any relation to the prevailing code rate and only indicates the number and size of TUs.

The header formats shown in Figure 4 are further described in **Table I.**

Coded MAC Header Formats for GMSK

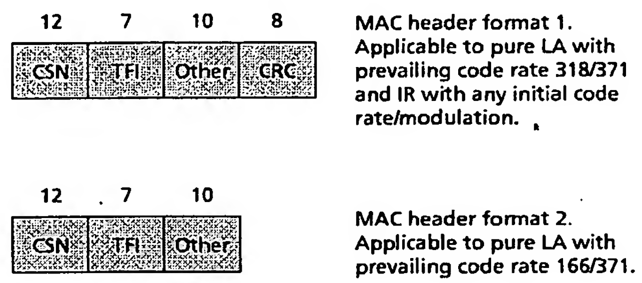
In both the GMSK cases, only one CSN needs to be indicated. There is no need for a CR field in this case, since the header format also indicates the code rate.

MAC header formats for GMSK are shown in **Figure 5** and further described in **Table II.** Header formats, which are identified through stealing bits, uniquely identify the code rate; therefore, there is no

Table I. MAC header formats for 8-PSK (header format 3 is used only when there are retransmissions with $D = 2$ incremental redundancy).

MAC header format (indicated through SBs)	MAC header	Number of CSNs	Header code rate (separately coded)
00	Short MAC header (39 bits)	1	39/76 (= 0.513)
01	Extended MAC header format 1 (51 bits)	2	51/129 (= 0.395)
10	Extended MAC header format 2 (63 bits)	3	63/129 (= 0.488)
11	Extended MAC header format 3 (118 bits)	7	118/235 (= 0.502)

CSN – Coded sub-block number
MAC – Media access control
PSK – Phase shift keying
SB – Stealing bit



CRC – Cyclic redundancy check
CSN – Coded sub-block number
IR – Incremental redundancy
LA – Link adaptation
MAC – Media access control
TFI – Temporary flow identifier

Figure 5.
MAC header formats for GMSK.

Table II. MAC header formats for GMSK.

MAC header format (indicated through SBs)	MAC header	Header code rate	LA code rate for data	Header coded with data	Remarks
00	Format 1 (37 bits)	37/73 (= 0.507)	318/371 (= 6/7)	No	Seamless adaptation with 8-PSK.
01	Format 2 (29 bits)	195/444 (= 0.439)	195/444 (= 0.439)	Yes	Two-step segmentation enabling robust mode; no IR in this mode.
10	Format 3 (GPRS CS-1)	GPRS CS-1	GPRS CS-1	Yes	Signaling option.
11	Reserved				

CS – Coding scheme
GMSK – Gaussian minimum shift keying
GPRS – General Packet Radio Service
IR – Incremental redundancy
LA – Link adaptation
MAC – Media access control
PSK – Phase shift keying
SB – Stealing bit

Table III. RLC block sizes and code rates when mapping one RLC block to each radio block.

Scheme	RLC block size (bits)	Code rate	Peak throughput (kb/s)
PCS-1	408	1/3	20.4
PCS-2	622	1/2	31.1
PCS-3	831	2/3	41.55
PCS-4	936	3/4	46.8
PCS-5	1,040	5/6	52.0
PCS-6	1,256	1	62.8

PCS – Puncturing and coding scheme
RLC – Radio link control

need for a separate CR field. The CRC is dropped in format 2 since it is coded along with the data.

Simulation Results

Simulations were carried out to examine the throughput tradeoffs between the following RLC segmentation approaches:

- Transmission of variable-length RLC blocks per radio block to achieve different code rates[3] (this approach results in the transmission of long RLC blocks at higher code rates); and
- Transmission of a variable number of short, fixed-length RLC blocks per radio block to achieve different code rates as proposed above.

The RLC block sizes and code rates for the variable-length RLC block approach are listed in **Table III**. All the coding schemes listed are derived by puncturing a rate 1/3, memory 6 convolutional code.

Simulation Conditions

Simulations were performed under the following conditions:

- Typical urban channel model as specified in GSM with a mobile speed of 3 km/h (TU3);
- Ideal frequency hopping and no frequency hopping;
- Only additive white Gaussian noise (AWGN) simulation—no interference (average carrier-to-noise ratio (C/N) is fixed for each simulation);
- Convolutional coding schemes derived from a rate 1/3, memory 6 convolutional code;
- Symbol rates and burst structure as specified in GSM;
- 1,392 bits per 4 GSM bursts (includes 8 stealing bits, 36 bits for USF, and 76 bits for coded MAC header);
- Identical coded MAC header and USF in all cases;
- 14-bit CRC assumed for short RLC blocks;
- No interleaving for uncoded transmissions with 1 RLC block mapped to each GSM burst in this case;
- For all code rates except unity, 4-block rectangular interleaving similar to SACCH interleaving in GSM;[6]
- Linearized GMSK pulse-shaping to decrease the peak-to-average ratio; and
- GSM training sequence, practical channel estimation, and reduced complexity equalizer.

Table IV. Throughput comparison for short and long RLC blocks with different 8-PSK code rates.

Approach	Peak throughput (kb/s)
Short, fixed-length RLC blocks	(304 X 4)/0.02 = 60.8
Variable-length RLC blocks	62.8

PSK – Phase shift keying
RLC – Radio link control

Table IV provides a throughput comparison for the short and long blocks with different 8-PSK code rates. The peak throughput for the short RLC blocks is smaller because of the CRC overhead.

Performance with Ideal Frequency Hopping

Figures 6 and **7** show the results with frequency hopping. Figure 6 shows the BLER performance as a function of the C/N for the following approaches:

- One large, uncoded RLC block/4 GSM bursts; and

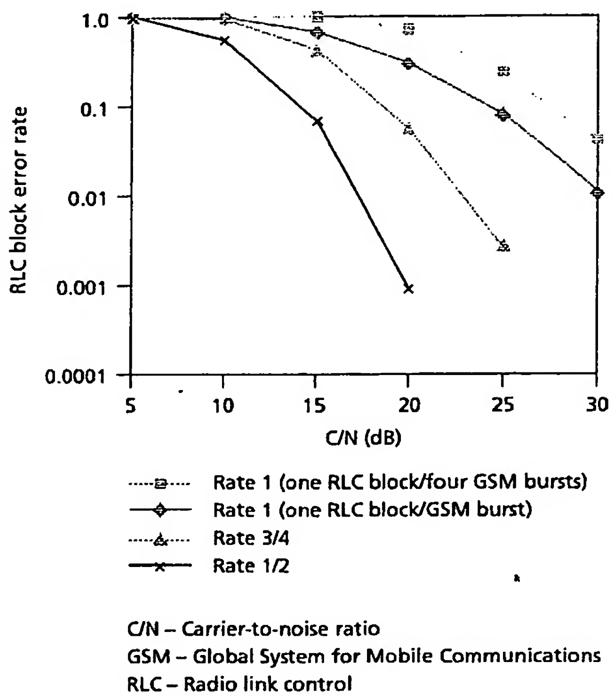


Figure 6.
BLER performance for short, fixed-length RLC blocks with 318 bits per block and ideal frequency hopping (TU3 channel model).

- Short, fixed-length RLC blocks with 318 bits/block (1 RLC block is mapped to each burst in the uncoded case).

In the uncoded case, the error rate for short blocks is significantly less than that for the large RLC block (at an error rate of 10%, the gain in C/N is approximately 3 dB). The gains are smaller without frequency hopping.

To study the impact of the short RLC blocks on the throughput, it is fruitful to compare the BLER gain (rather than the dB gain) with the overhead. The additional overhead for the short RLC blocks is in the range of 4 to 5%. When the operating code rate is such that the difference in BLER between the long and short blocks exceeds 4 to 5%, the scheme with short blocks can achieve higher throughput.

The BLER simulations were used to estimate the throughput performance of IR for each initial code rate. Header errors were taken into account in the throughput computation; the BLER and throughput results assume the use of rate 1/2 encoding on the header across all initial code rates. In the short RLC block case, extended headers are used when retransmissions which are not in sequence are included. The extended header translates into a small reduction in the amount of soft information available from IR parity transmissions. This reduction is taken into account in the throughput computation. The first transmission of each RLC block occurs at the initial code rate. Subsequent retransmissions provide incremental soft information that may be combined with previously stored soft information in order to decode the RLC block.

Figure 7a shows the throughput performance for IR ($D = 1$) with short, fixed-length RLC blocks and different initial code rates. The throughput results do not show any distinct code rate switching points. However, different code rates are required in order to achieve acceptable delay performance, since the BLER for higher initial code rates is large at low C/N. The code rate switching points may be determined using delay thresholds.

Figure 7b shows the throughput comparison between the short and long blocks. $D = 2$ IR with short

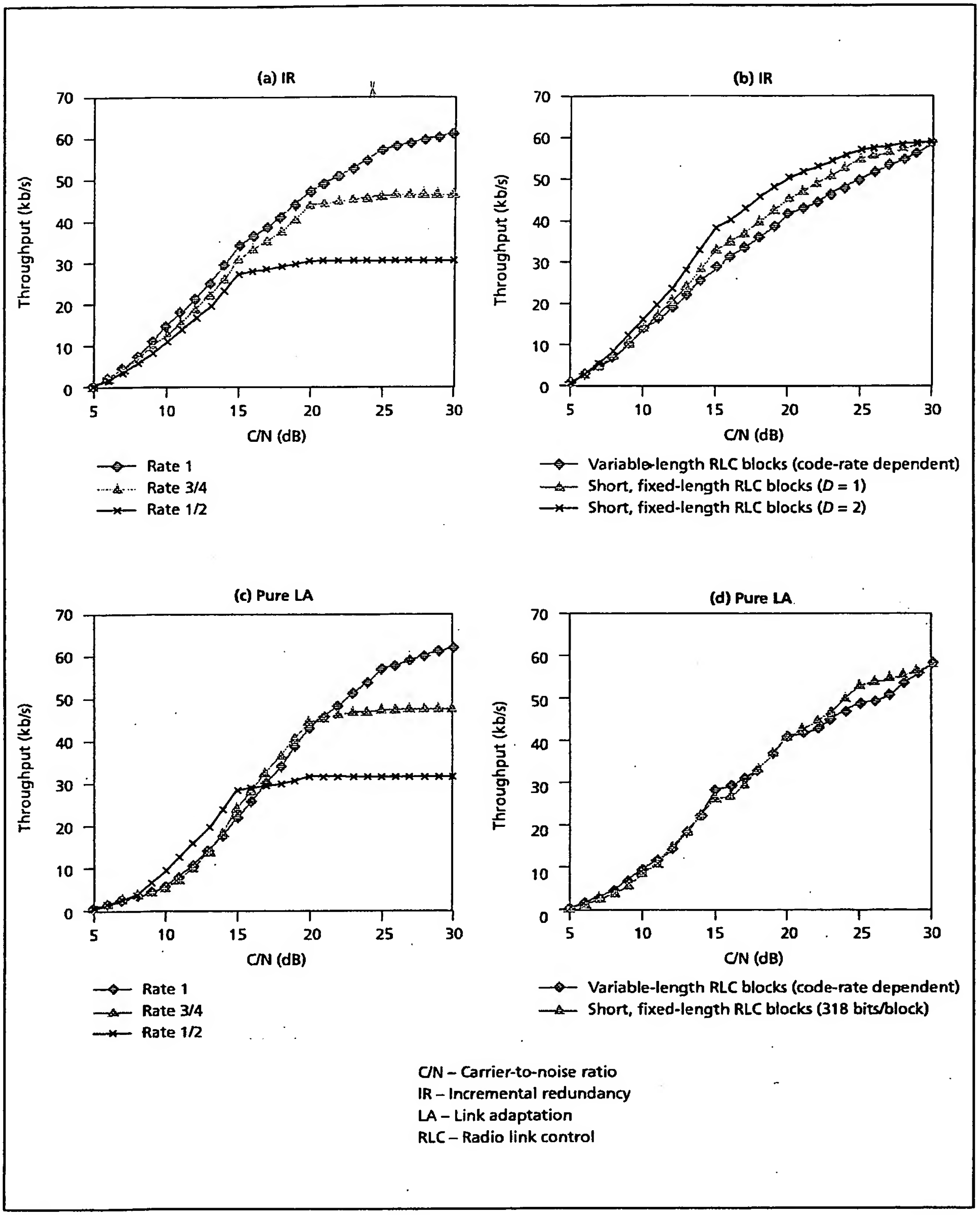


Figure 7.
Throughput performance with ideal frequency hopping and different initial code rates (TU3 channel model).

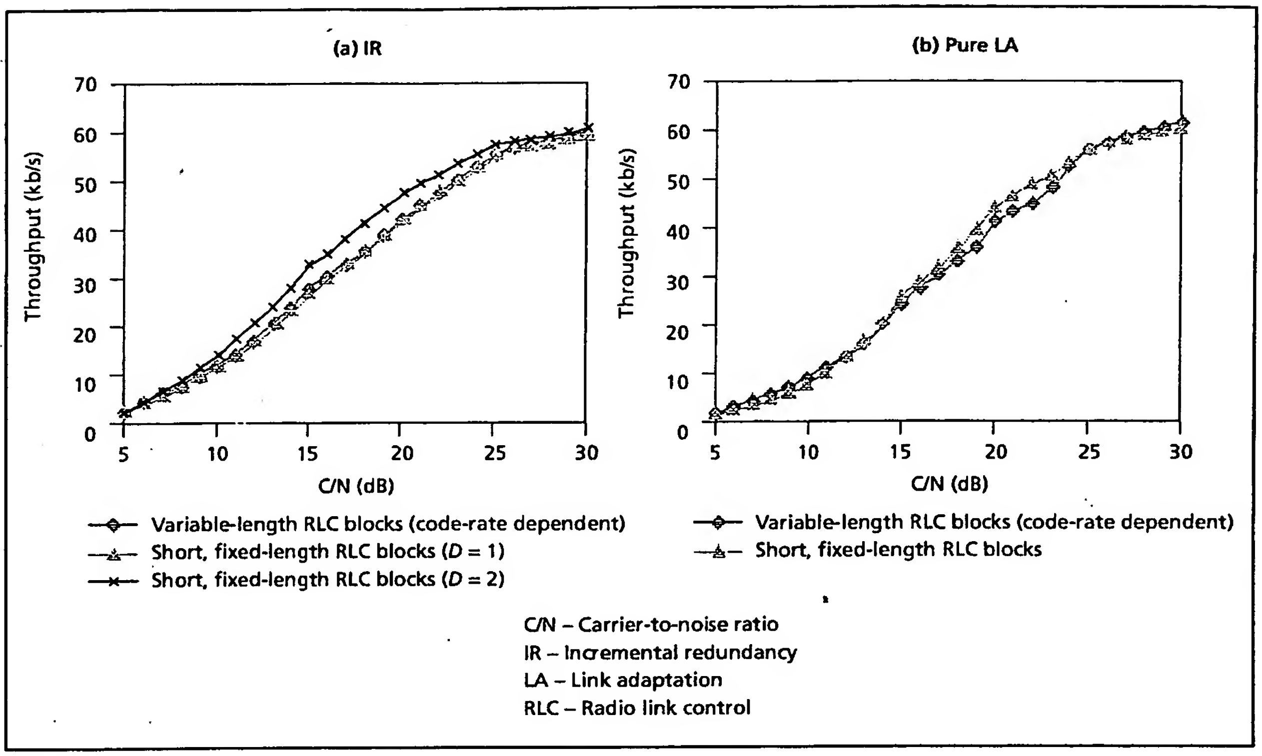


Figure 8.
Throughput performance with no frequency hopping and different initial code rates (TU3 channel model).

blocks achieves the highest throughput, followed by $D = 1$ IR for short blocks. At 20 dB, the gain in throughput with short blocks is approximately 10% for $D = 1$ and 20% for $D = 2$. The gain in throughput is primarily due to the large performance improvement for uncoded transmissions. The results show that for all C/N values *below 29 dB*, the short, fixed-length RLC block approach achieves higher throughput. Above 29 dB, the large RLC blocks achieve higher throughput than the short RLC blocks because of the difference in overhead.

The system exhibits similar behavior when only code rate adaptation is carried out without any soft combining at the receiver (the pure LA case). Figure 7c shows the throughput performance with short, fixed-length RLC blocks and different code rates. The results account for the fact that in the uncoded mode, retransmissions are carried out at rate 3/4 when the retransmitted blocks are not in sequence. In these cases, distinct code rate switching points are visible from the throughput results; in reality, however, different code rate switching points may be determined by setting C/N thresholds corresponding to delay budgets.

Figure 7d shows that for C/N values between 20 and 30 dB, short RLC blocks offer 5 to 10% higher throughput. Due to higher code rate granularity, the larger blocks achieve higher throughput between 14 and 18 dB. We observe that three 8-PSK coding schemes are sufficient to achieve throughput benefits across the range of C/N values.

Performance with No Frequency Hopping

When no frequency hopping is used, the short RLC blocks achieve lower error rate for all coding schemes, but the gain in the uncoded case is not as large as that achieved with frequency hopping. The improvement in the uncoded case is approximately 1 dB at an error rate of 10%.

Figure 8 shows the comparison between the short and long block approaches for both IR (Figure 8a) and pure LA (Figure 8b). Again, in the IR case, it is observed that $D = 2$ IR with short blocks achieves the highest throughput; the gain is approxi-

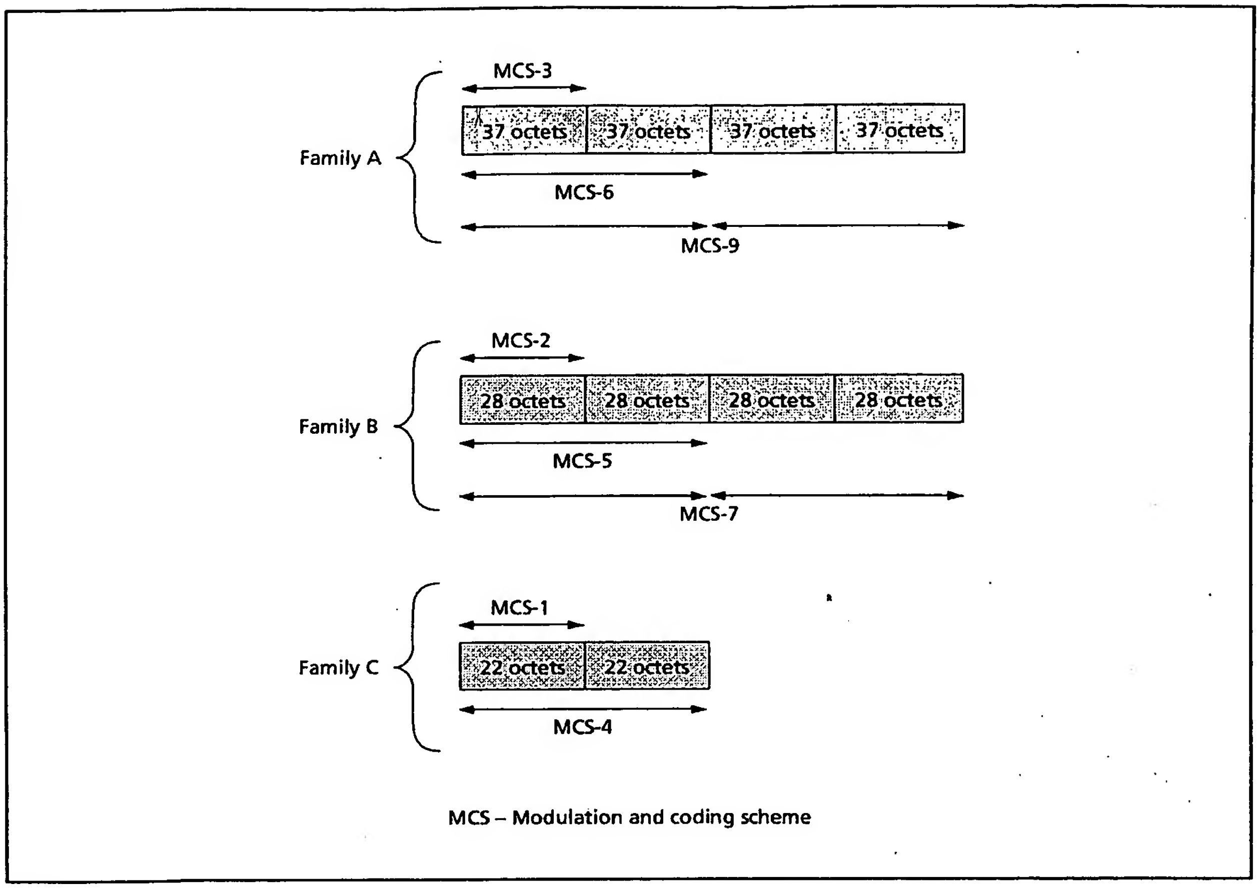


Figure 9.
Relationship of the three RLC block sizes to the nine modulation and coding schemes in EGPRS.

mately 10% in the range 13 to 23 dB. The throughput performance for $D = 1$ IR is similar for both short and long blocks below 25 dB; above 25 dB, the long blocks achieve higher throughput. For pure LA, the short blocks offer around a 5% throughput gain below 25 dB, while the long blocks achieve higher throughput above 25 dB.

Summary of Simulation Results and Conclusions

The results of the simulations are as follows:

- For IR, short RLC blocks offer 10% higher throughput with frequency hopping and comparable throughput without frequency hopping.
- The scheme permits a simple extended header format 3 that permits $D = 2$ IR operation with little additional complexity. $D = 2$ IR offers additional 10% throughput gain in both the frequency hopping and no frequency hopping cases.
- For pure LA, the difference in throughput between the schemes using short and long blocks is small.
- With frequency hopping, IR with $D = 1$ offers a 10% gain over pure LA. Without frequency hopping, IR with $D = 1$ and pure LA offer comparable performance, while IR with $D = 2$ offers 10% higher throughput.

Based on the results of this study, the following conclusions are drawn:

- Segmentation of the LLC data into short, fixed-length RLC blocks achieves higher throughput over a wide range of operating conditions compared to an alternative approach in which the RLC block size is larger and varies with the code rate.

Table V. EGPRS modulation and coding schemes with peak data rates.

Scheme	Modulation	Maximum rate (kb/s) per slot	Code rate	Header code rate	Blocks per 20 ms	Family
MCS-9	8-PSK	59.2	1.0	0.35	2	A
MCS-8		54.5	0.92	0.35	2	A
MCS-7		44.8	0.76	0.35	2	B
MCS-6		29.6	0.49	1/3	1	A
MCS-5		22.4	0.37	1/3	1	B
MCS-4	GMSK	17.6	1.0	1/2	1	C
MCS-3		14.8	0.80	1/2	1	A
MCS-2		11.2	0.66	1/2	1	B
MCS-1		8.8	0.53	1/2	1	C

EGPRS – Enhanced General Packet Radio Service
GMSK – Gaussian minimum shift keying
MCS – Modulation and coding scheme
PSK – Phase shift keying

- The improvement in BLER performance for short blocks along with LA and IR more than compensates for the increase in overhead. The small loss in throughput at high C/N values is acceptable, since the majority of mobile stations will not operate at such high C/N values.
- The scheme with short, fixed-length RLC blocks permits seamless LA without any loss of throughput while switching code rates.
- Three coding schemes (rates 1, 3/4, and 1/2) are sufficient in order to achieve the throughput benefits of code rate switching over typical operating conditions. The advantages of higher code rate granularity are not significant when IR is the primary mode of operation; moreover, it is difficult to determine adaptation thresholds that remain valid for different mobile environments.
- Frequency hopping offers better throughput performance across the range of carrier-to-interference-plus-noise values of interest.
- The aforementioned segmentation approach enables the support of IR with $D = 2$, providing a clear throughput advantage as shown in the results.

Summary of EGPRS Radio Link Control Standard

RLC block sizes and segmentation schemes have been designed based on the joint proposal by several companies to ETSI. The chosen scheme employs $D = 1$, reducing implementation complexity by fixing the number of RLC block decoding operations required per 20 ms. As a tradeoff between overhead and BLER, at the highest rates, the block size is chosen to accommodate two RLC blocks per 4 GSM physical layer bursts. For seamless adaptation, one block can be accommodated at a lower coding rate within the same family.

Modulation and Coding Schemes and RLC Block Sizes

The joint proposal defines nine modulation and coding schemes (MCS-1 to MCS-9): 4 code rates with GMSK and 5 code rates with 8-PSK. MCS-9 is uncoded 8-PSK. Due to concerns about the performance of uncoded 8-PSK, MCS-8 has been specified with very light coding using a code rate of 0.92. It has been shown that in the presence of phase noise, power amplifier nonlinearities, and adjacent channel interference, MCS-8 (encoded at rate 0.92) outperforms MCS-9. However, MCS-9 provides the highest peak rate when future radio frequency (RF) hardware improvements are in place.

Three RLC block sizes are defined for the nine modulation and coding formats shown in **Figure 9**. This is done to facilitate retransmissions across varying SINR. According to the link quality, an initial MCS is selected for an RLC block. For the retransmissions, the same or another MCS from the same family of MCSs can be

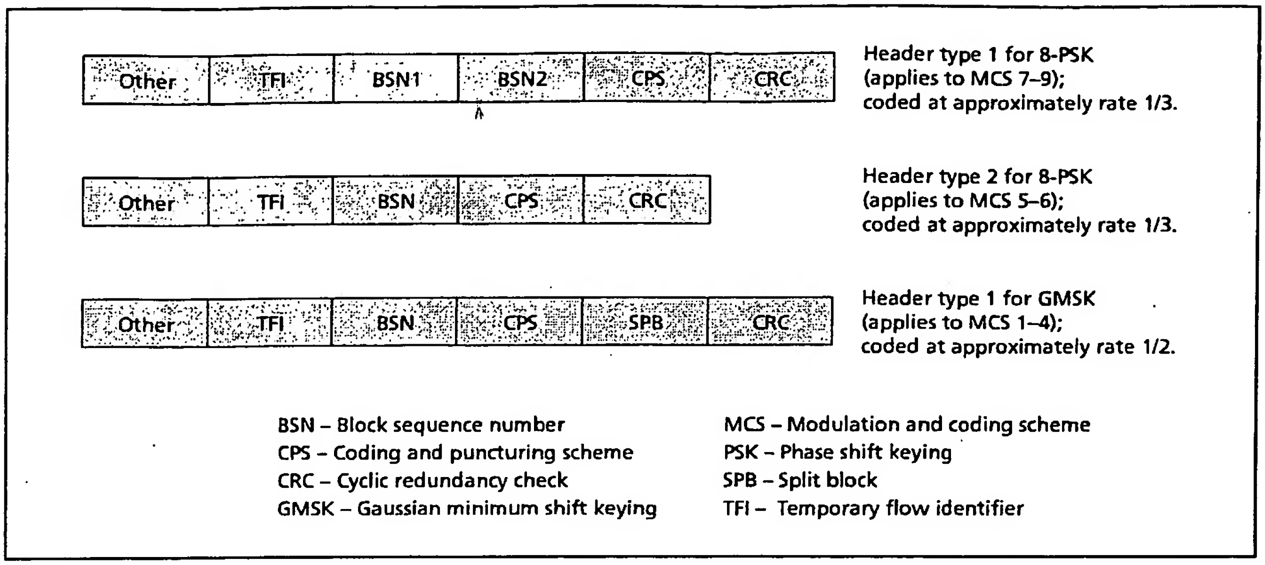


Figure 10.
Header formats for LA and IR as a function of the prevailing MCS (applies to both the downlink and the uplink).

Table VI. Definitions of fields included in the different header formats.

Field	Field size (bits)	Function
TFI	5	Identifies the TBF to which the RLC data blocks within the radio block belong.
BSN1	11	Block sequence number of the first RLC data block.
BSN2	10	Incremental block sequence number of the second RLC data block relative to the first RLC data block.
CPS	5 for header format 1 3 for header format 2 4 for header format 3	Indicates channel coding and puncturing format used for RLC data blocks.
SPB	2	Only applies to header format 3; indicates if resegmentation has been carried out for retransmission of the RLC data block.
Other	6 for downlink 9 – 17 for uplink	Includes spare bits and other fields defined in GPRS for managing the RLC/MAC layer.
CRC	8	Provides error detection capability.

BSN – Block sequence number
CPS – Coding and puncturing scheme
CRC – Cyclic redundancy check
GPRS – General Packet Radio Service
MAC – Media access control
RLC – Radio link control
SPB – Split block
TBF – Temporary block flow
TFI – Temporary flow identifier

selected. For example, MCS-9 carries two RLC blocks, each of size 74 bytes. If the SINR deteriorates, the 74-byte RLC blocks may be retransmitted using MCS-6 (8-PSK encoded at rate 1/2) with one block per four GSM physical layer bursts. If additional coding is required, the 74-byte RLC data block can be further segmented into two 37-byte sub-blocks, and each can be transmitted using MCS-3 (GMSK with code rate 4/5). The header should indicate that this is a segmented portion of a 74-byte RLC block and not a new transmission

using 37-byte blocks. Thus EGPRS provides plenty of flexibility for block-by-block rate adaptation.

EGPRS allows seamless transition between pure LA and IR depending only on receiver memory. The data block is first encoded using a rate 1/3 convolutional code. Puncturing schemes P1, P2, and P3 are defined to puncture the rate 1/3 code to the desired code rate. Following the proposal of Kallel,[7] the individual puncturing schemes are designed to be as disjunctive as possible in order to achieve good performance with IR. In addition, the puncturing schemes are designed to ensure that the individual schemes (P1, P2, and P3) achieve comparable error performance for pure LA.

At the operating MCS, the initial transmission of a block consists of the bits obtained by applying puncturing scheme P1 (for the chosen MCS) to the rate 1/3 encoded data. On receiving a negative acknowledgment for the RLC block, additional coded bits (that is, the output of the rate 1/3 encoded data punctured with scheme P2 of the prevailing MCS) are transmitted. If all the punctured versions of the encoded data block have been transmitted, the cycle is repeated starting again with P1. If the receiver does not have sufficient memory for IR operation, it can attempt to decode the data by using the information corresponding to just P1, P2, or P3. This corresponds to the pure LA case. For IR operation, the receiver must have sufficient memory in order to store soft information corresponding to RLC blocks that have not yet been decoded successfully. Each time the receiver obtains additional coded bits, it attempts soft decision decoding using this redundant information in addition to previously stored soft information corresponding to the same RLC block(s).

Table V shows the coding rates, modulation schemes, and peak data rates for EGPRS.

Header Formats

Separately coded headers are used as required for IR. The header is robustly coded so that the receiver is able to determine the block identities for all transmissions, even if the payload cannot be decoded. The same headers and transmission formats are used for both LA and IR. **Figure 10** shows the different header formats. Three header formats are defined and the header format is indicated through stealing bits included in each GSM burst. Coding schemes, data formats, and one or two block sequence numbers are included in the header.

Table VI provides a summary of the different fields employed in the header. The coding schemes for all the header types are derived by appropriately puncturing a rate 1/3, memory 6 convolutional code. Tailbiting is employed in header codings to save on overhead.

The receiver relies on blind detection to determine the modulation scheme for each radio block. Once the modulation is known, the header formats may be identified by decoding the stealing bits. For 8-PSK, 8 stealing bits are used allowing the reliable identification of 4 header formats. For GMSK, only one header format is currently defined, but 12 stealing bits are available to allow more header formats to be defined in the future.

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